BizPBX

Server Guide

Version: 2.4.0.2

(English Version)

Yeastar Technology Co., Ltd.

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1. Introduction

You may skip section 1- Introduction and directly move to section 2- Before you Proceed. However, we recommend reading through the introduction to familiarize yourself with the features and functions of BizPBX and help improve its operation. Thank you for purchasing BizPBX!

BizPBX —Unified Communication PBX

BizPBX is Windows-based software PBX solution, which is the world first communication system that realizes the free flexible externally connection with three popular networks Skype, telecom PSTN and SIP; internally work with analog phone, SIP phone, Skype mobile phone and computer softphone (can be used as an extension).
It not only includes complete and rich functions of PBX, but also integrates Skype server, internal Instant Messaging server and E-Mail server for easily realizing high efficiency and free communication.

Who Suit BizPBX?

BizPBX is a unified communication PBX especially suit for those SMBs who want to get the maximum functions at the minimum costs on company internal and external work. Using BizPBX, you can save much time and energy to manage your calls and also costs in purchasing different kinds of servers, such as file server, phone system, conference server and record server.

Why Choose BizPBX?

Small system, big features

BizPBX includes an amazing range of sophisticated, user-friendly features that give you complete call control. Hold, transfer and conference calls? Of course. Voicemail, auto attendant, music on hold, ring groups? Sure. But you also get advanced features like voicemail to E-mail, call recordings, remote extensions, instant message, and short message, and much more.

Flexible and communication system-PBX hybrid

BizPBX supports traditional and VOIP networks (SIP & Skype), so you choose how you want to connect. You can take advantage of VoIP and traditional telephone network service provider offerings and in the meantime enjoy the smooth, quick and even free communication (writing and speaking) through Call Messenger.

1.1 Benefits

1. Increase communication efficiency

- Internal Directory at a glance, online status, click-to-call.
- Customer Directory at a glance, ready for search, click-to-call.

- Initiate a conference call at any time, easy to use.
- Internal IM transmission in LAN & real-time arrival.
- Internal E-mail transmission in LAN & real-time arrival.
- Caller file bomb display, be prepared before answering a call.
- Call history available for search and push to call back.
- Retroactive recording to hold back important calls.
- Work at home as work in company office.

2. Save on call costs

- Call through VoIP/Skype.
- Calls among branches can be free.
- Access into BizPBX remotely to make free calls.

3. Improve corporate image & degree of satisfaction

- Increase call-in mode that provide web call-me as an option.
- IVR customizing function to foster a good company image.
- Ring group allows calls to be accepted as soon as possible.
- Receive fax in real-time to show the high efficiency.

4. Strengthen internal management

- Auto-record for all phone calls or designated ext.'s calls.
- Call record keep for future reference.
- Fax files retain.
- Network disk management for data security and backup

1.2 Features

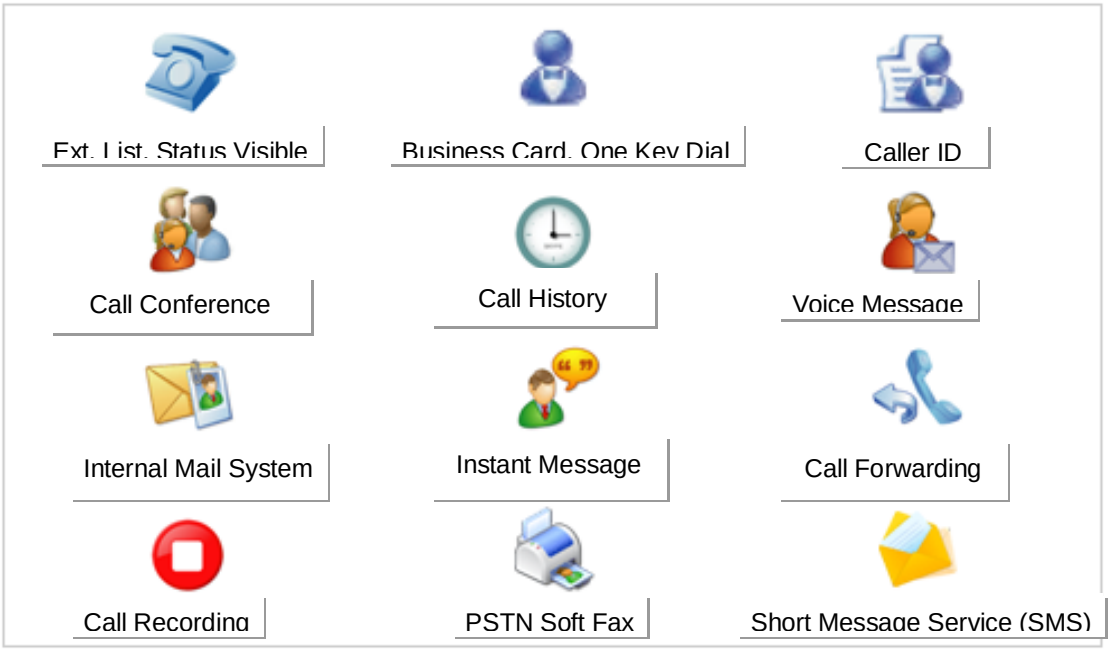
Automated Attendant	CallerID
Blind Transfer	Database Integration
Call Booking	Dial by Name
Call Detail Records	Direct Inward System Access(DISA)
Call Forward	Do Not Disturb
Call Monitoring	Extension Group
Call Parking	Interactive Voice Response(IVR)
Call Queuing	Local and Remote Call Agents
Call Recording	Music On Hold
Call Restriction	Protocol Conversion
Call Retrieval	Remote Call Pickup
Call Routing	Remote Office Support
Call Transfer	Roaming Extensions
Call Waiting	Share Contacts
Call Remind	Interoffice Trunking

1.3 Minimum System Requirement

- PC Requirement
  Intel Core Duo & 2G RAM for 64 Simultaneous calls & 4 Skype trunks;

- Operating System
  Windows XP Professional + SP2
  or
  Windows Server 2003 + SP2

- Internet Connection
  Different numbers of Skype concurrent calls require different bandwidth; each port average occupies 3Kb/s to 16 Kb/s.



2. Before You Proceed

Note the following important precautions before you install the software.

1. Choose a computer with suitable configuration to run as the server (referenece on 1.3)
2. Install OS Windows XP Professional SP2 or Windows Server 2003 SP2.
3. Don't use the clone way to install the operating system of BizPBX Server PC.
4. Make sure BizPBX Server PC designated for BizPBX has a clean system with only anti-virus installed.
5. Make sure BizPBX Server PC designated for BizPBX has no Skype applications installed.
6. Make sure BizPBX Server PC has no SIP Server or SIP Softphone installed.
7. Make sure BizPBX Server PC designated for BizPBX has no domain controller installed and the disable the password complexity.
8. Make sure BizPBX Server has a dedicated broadband access to ensure voice quality.
9. Make sure to login on BizPBX Server with an account that has administrator privileges.
10. Please don't login BizPBX server by "Remote Desktop Connection", but use VNC in case you need remote login BizPBX server.

3. Installation Procedure

Below is BizPBX installaton and configuration flow diagram that gives you an overview of all the steps required in installing BizPBX.

BizPBX Installation Flow

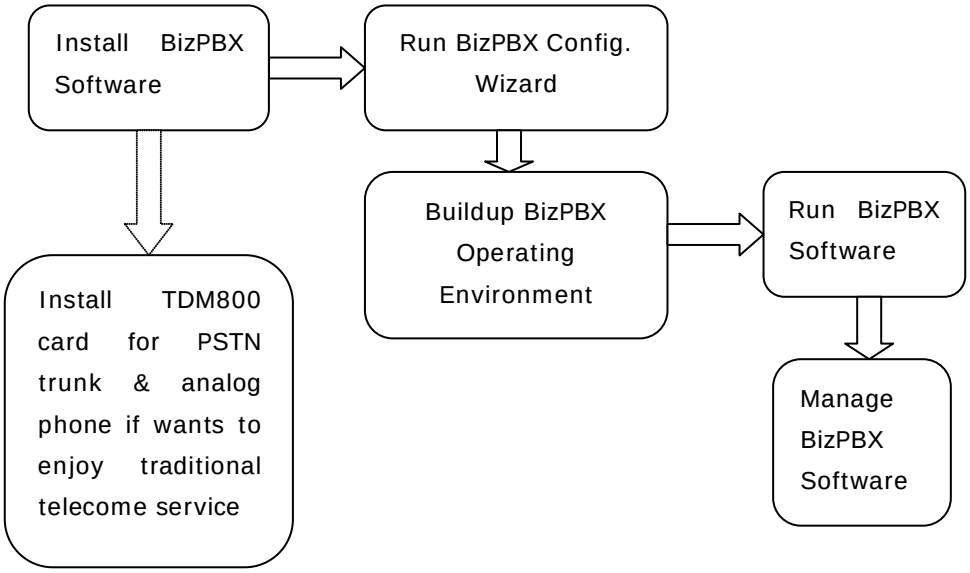


Figure 3-1

4. Installing BizPBX Software

This section shows how to install BizPBX software on the PC.

1. Download the BizPBX from:
   http://www.yeastar.com/download/all/All.asp.
2. Doube-click the 'BizPBXSetup.exe', Welcome to the BizPBX Installation screen will come up. Click Next to continue. See Figure 2.

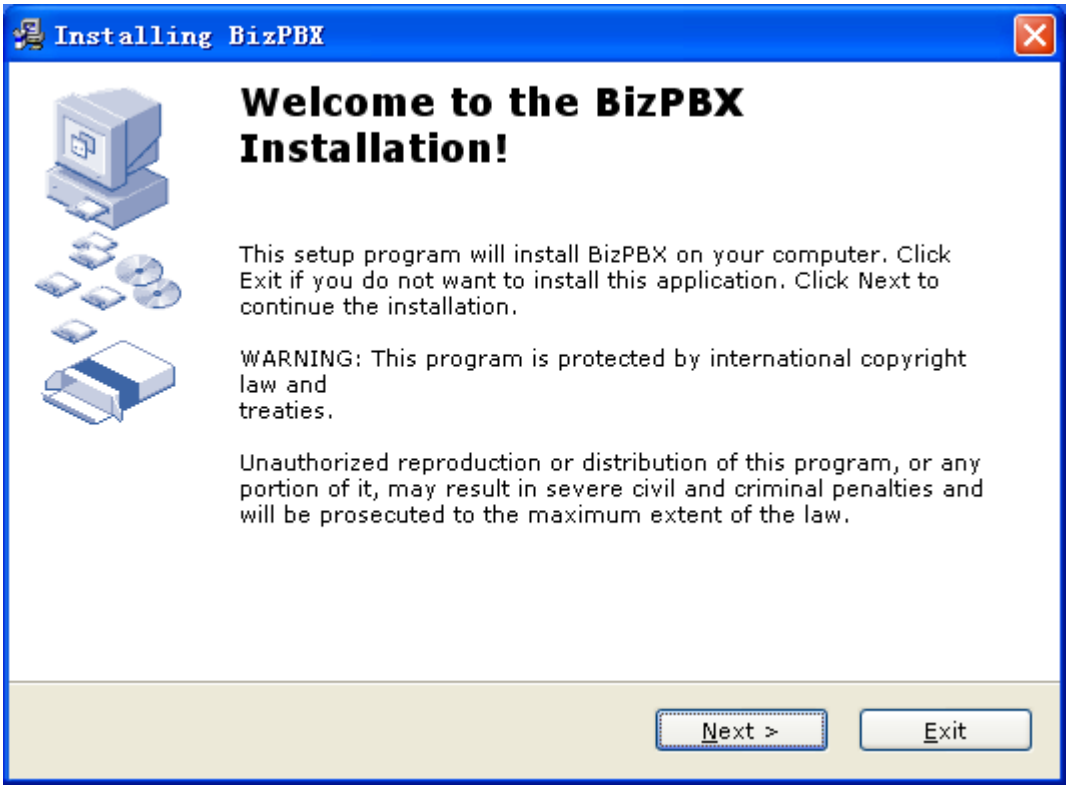


Figure 4-1

3. Read through the software License Agreement, select 'I agree with the above terms and conditions', and then click **Next** to continue. See Figure 3.

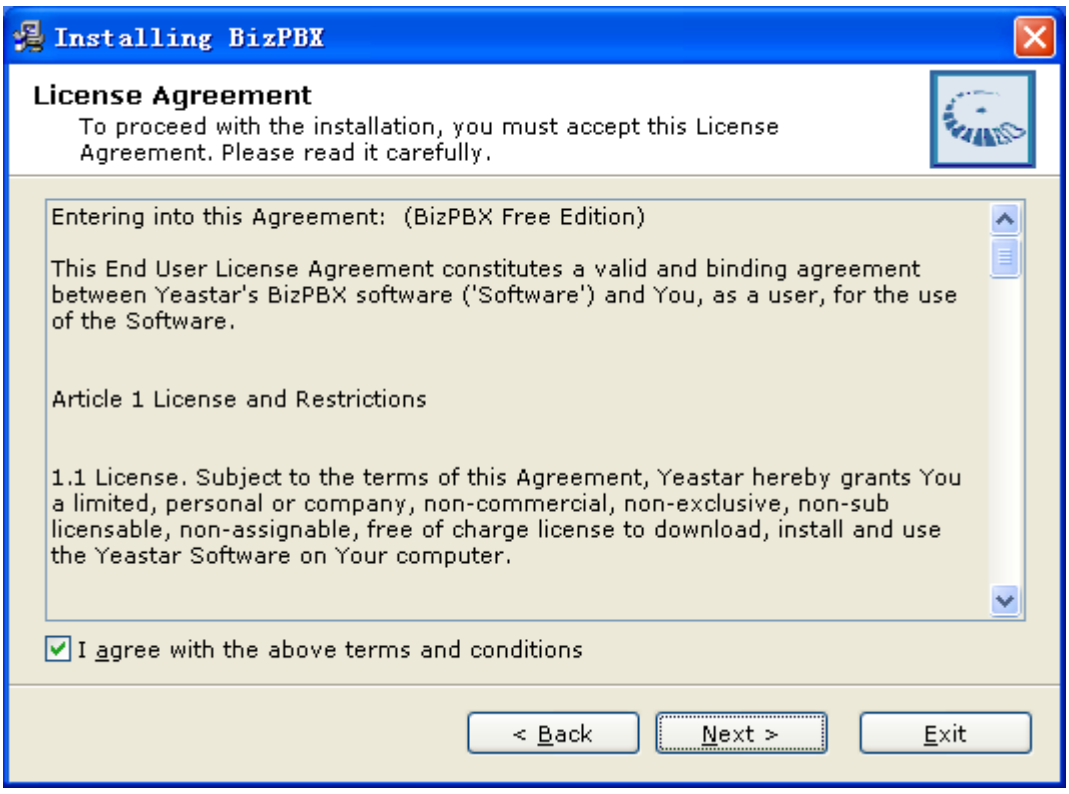


Figure 4-2

4. An info window will appear to remind you that **if you want to use PSTN line or desk analog phone**, BizPBX needs the dedicated analog interface hardware card TDM800 supplied by Yeastar.

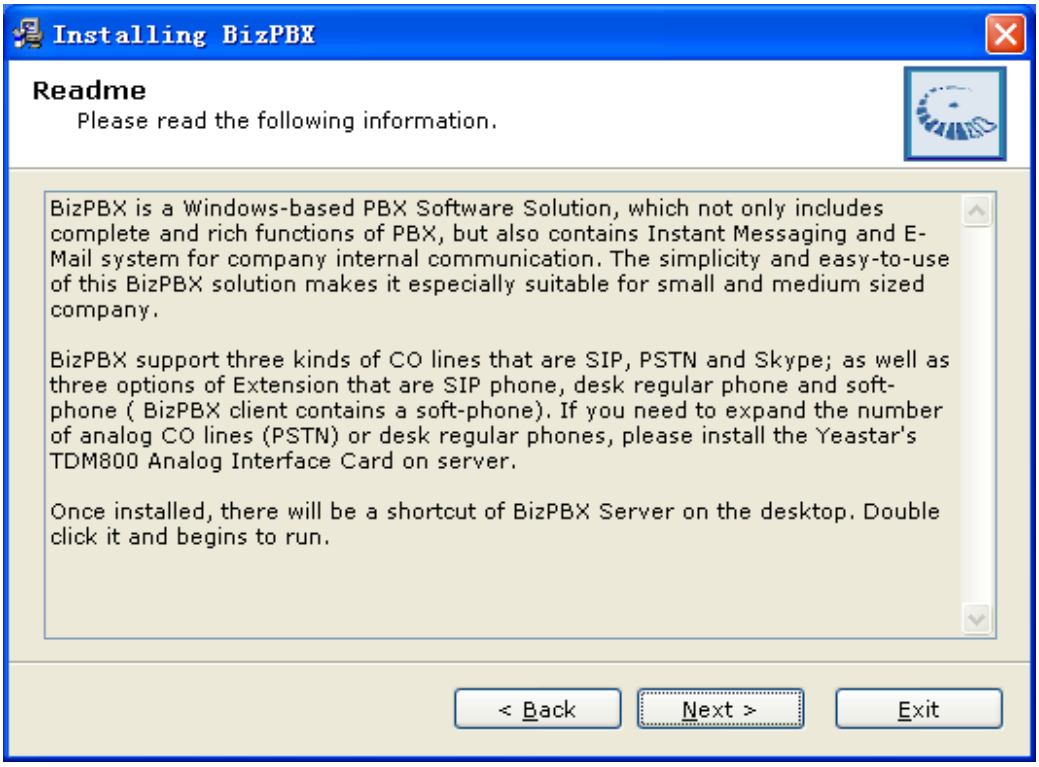


Figure 4-3

5. The Destination folder window will offer you the option where you would like BizPBX to be stored on your computer. Click Next to continue.

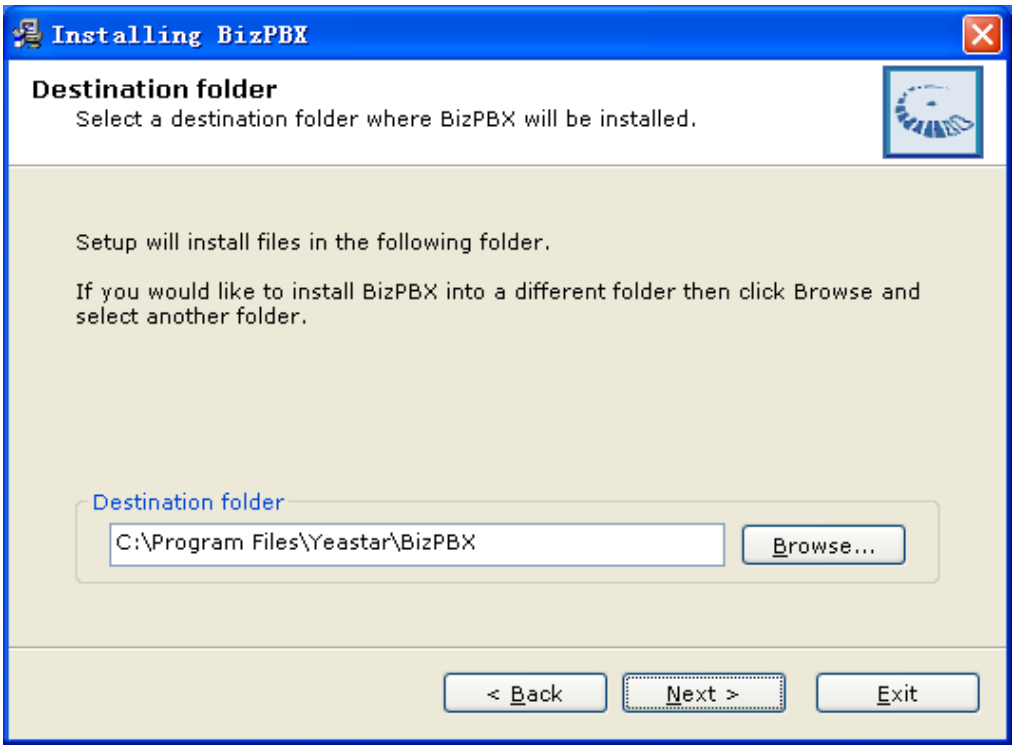


Figure 4-4

6. Enable the options by your own demands, and then click Next.

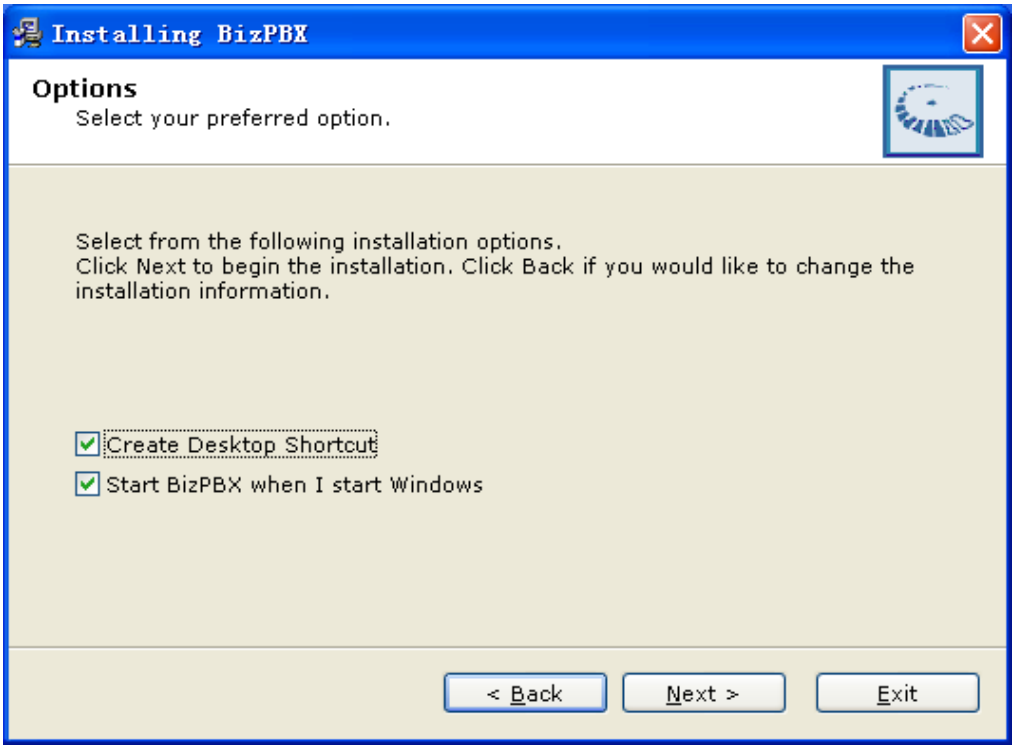


Figure 4-5

7. Enter into the Installing Files，system begin to configuration, which will

last for a while.

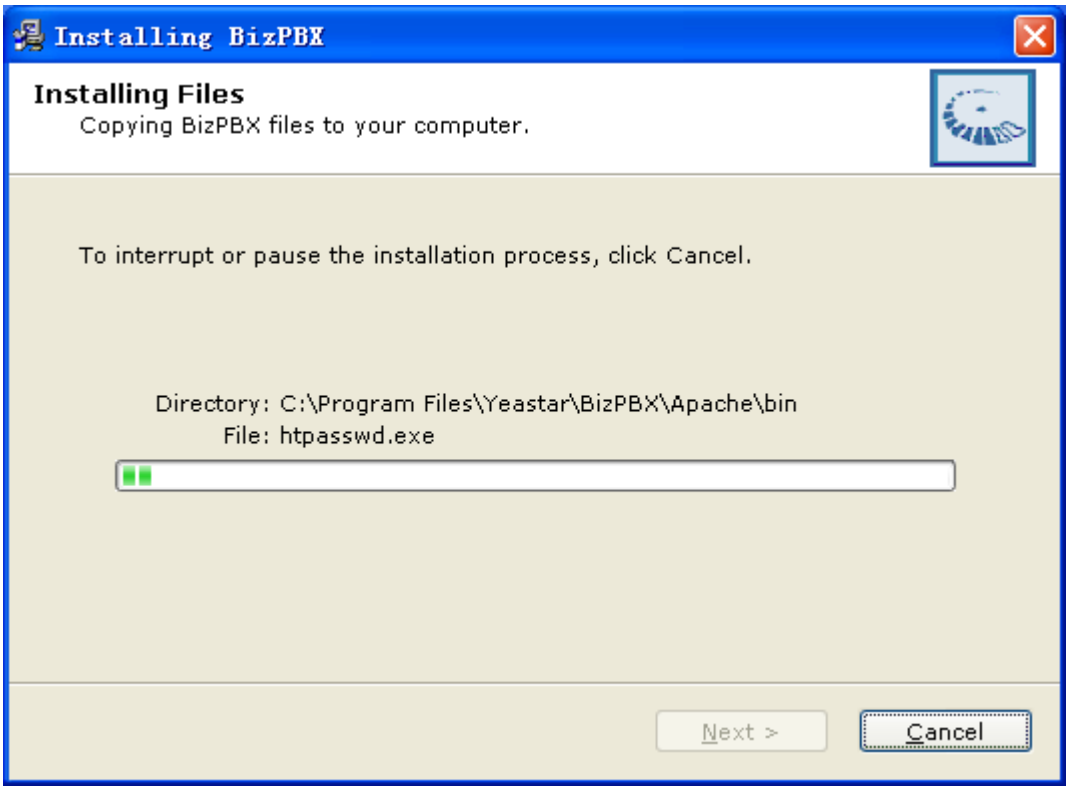


Figure 4-6

8. When the installation is complete, a screen pop-up to notify you that the software is installed successfully. Click Finish.

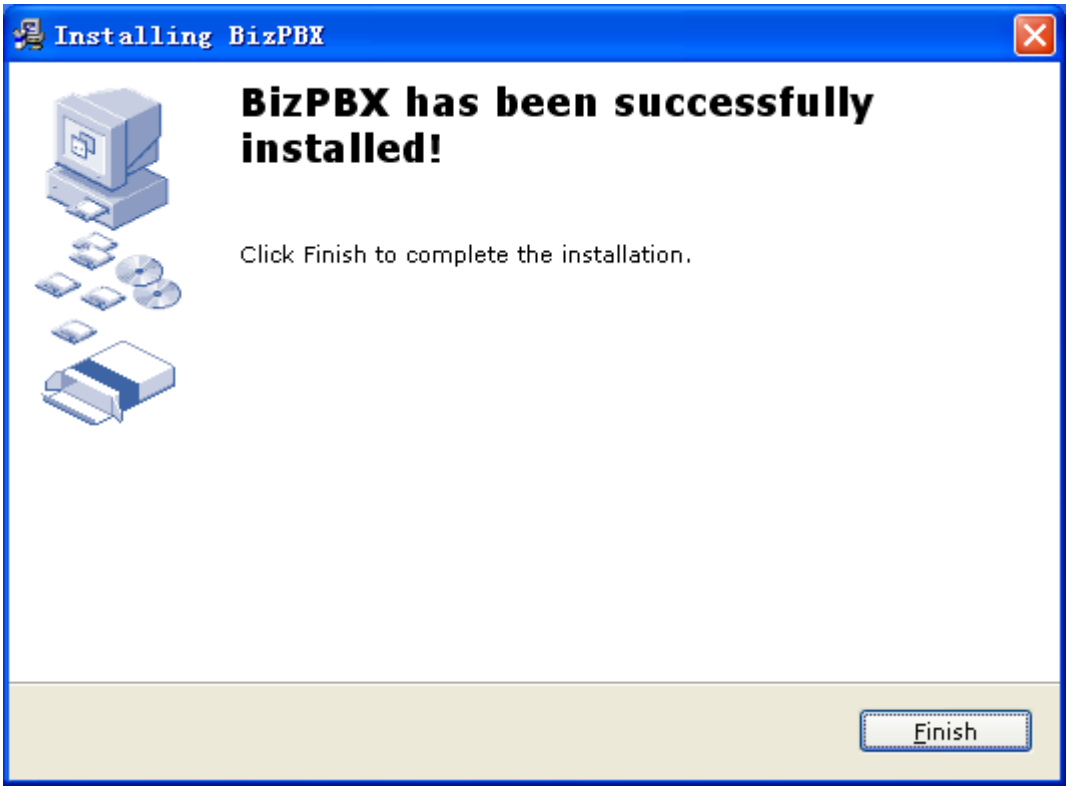


Figure 4-7

9. The final screen reminds you to restart computer in order to complete the

installation. You would better to restart now.

Figure 4-8

10. Launch BizPBX to enter the next chapter.

Note: Before installing BizPBX software, you should first uninstall Skype software on this BizPBX server installed computer if you already have Skype software installed.

5. Running Config Wizard

Configure BizPBX through Wizard.

You can click the 'Config' on the prompted menu from the BizPBX on the right foot of the screen to start the Config Wizard. If the 'Config' key is invalid, please click 'Stop' first to stop running BizPBX.

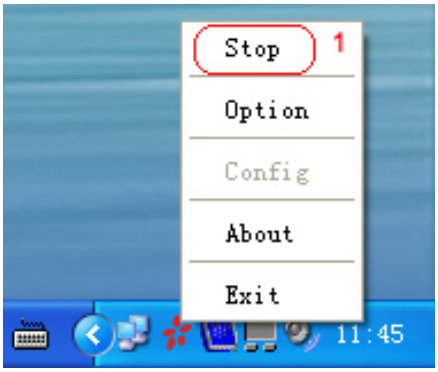


Figure 5-1

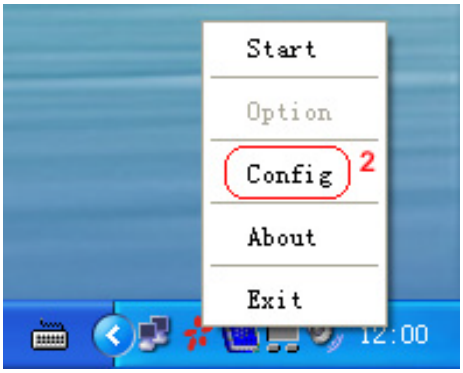


Figure 5-2

5.1 Launch the Config Wizard

Click Next

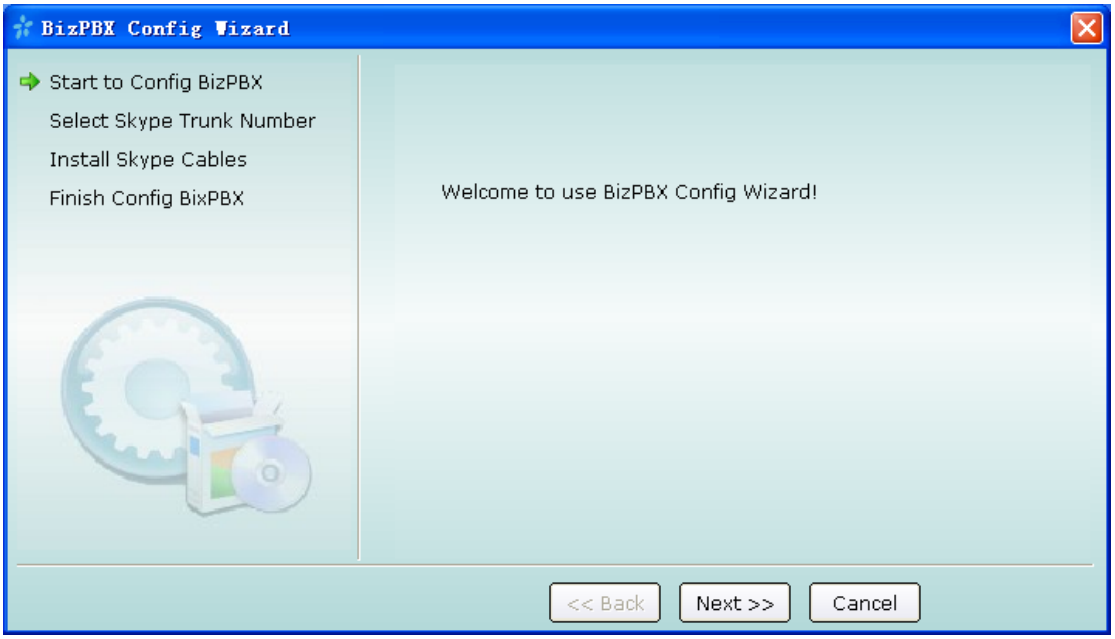


Figure 5-3

5.2 Choose Skype Trunk Quantity

Choose the Skype trunk quantity

Note: Please don't enable 'Install TDM800 Driver' if you don't connect PSTN line or analog regular phone.

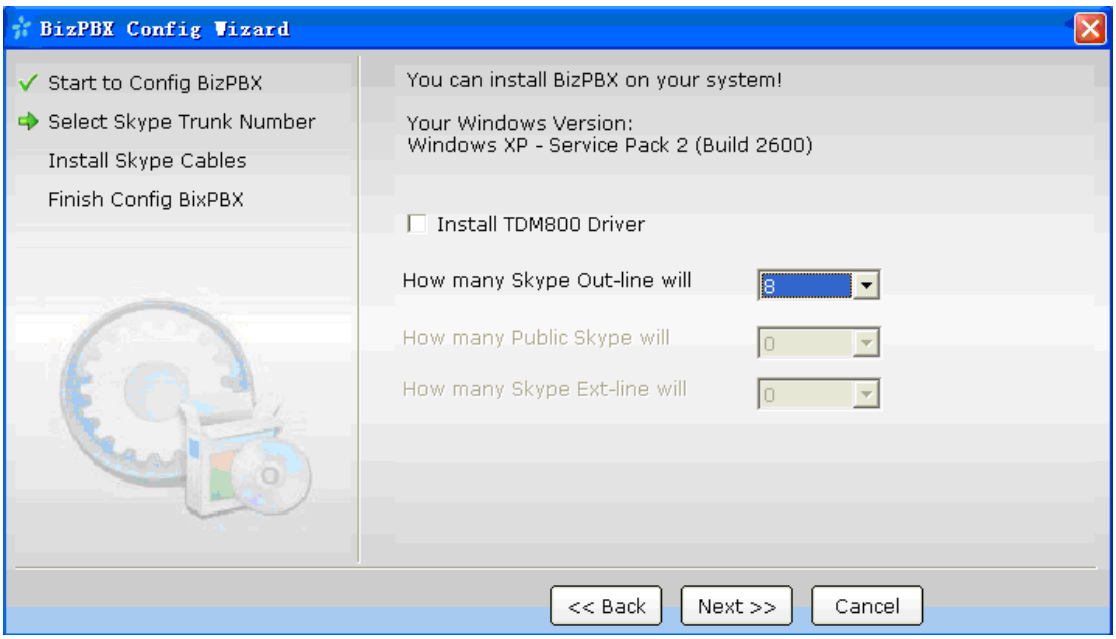


Figure 5-4

The following window maybe will appear during the system installation. Click

Continue Anyway

Figure 5-5

5.3 Configure Skype for Each Port

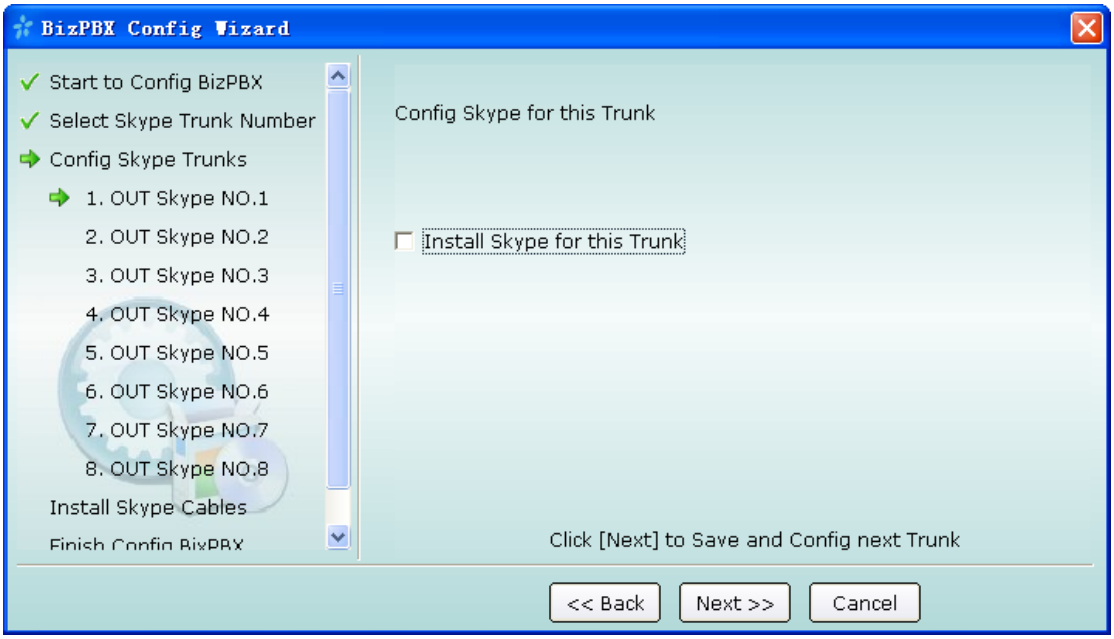


Figure 5-6

A Skype?—Creat Account will appear. Create a new account (see Figure 16) or cancel it and log in by using an existing Skype account.

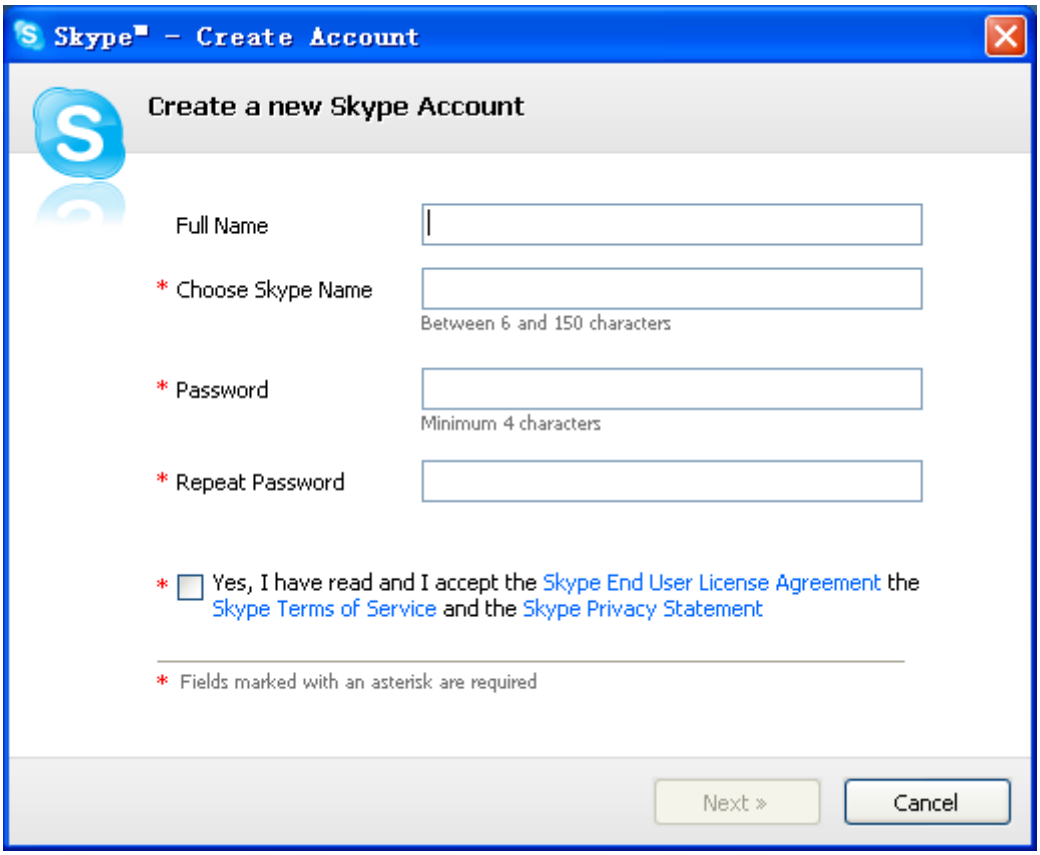


Figure 5-7

Enable the Sign me in when Skype Starts and wait for Skype to log you in successfully.

Figure 5-8

When BizPBX get the Skype Name automatically, the configuration of Port 1 is finished. Click Next to configure other ports by the same way.

Note: After these steps are complete, repeat **step 5.3** to configure the remaining ports and their Skype accounts. When the remaining ports are congifured, there will be a green tick before every port. See Figure 18

5.4 Install Skype Cables

Enable Install Skype Cables and click Next to continue.

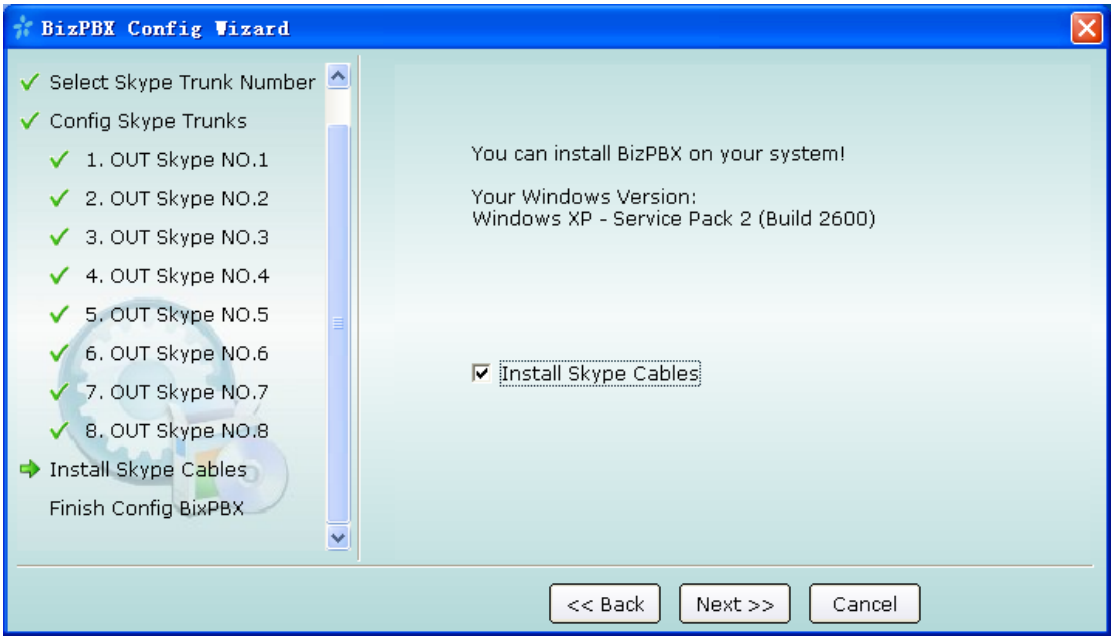


Figure 5-9

The following window maybe will appear during the system installation. Click Continue Anyway

Figure 5-10

5.5 Finish Config Wizard

Choose the country where you are, and the saving path of sound file. Suggest you to not change the default of flash time settings.

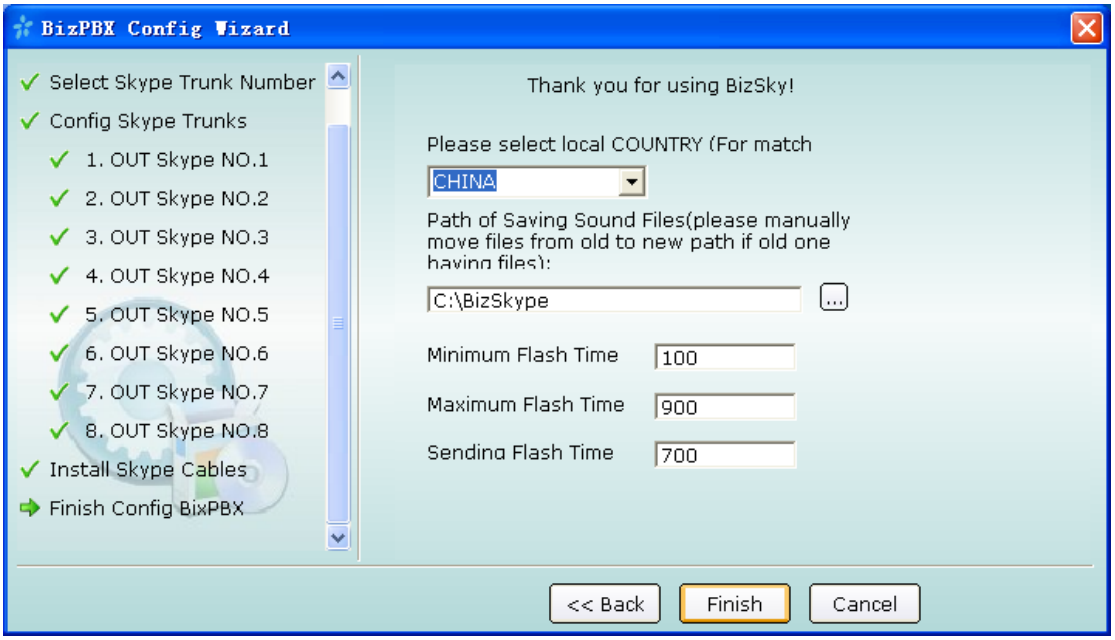


Figure 5-11

BizPBX system configuration is complete!

6. Running BizPBX Software

Step 1 Double-click the shortcut on desk to run the software

Figure 6-1

Step 2 Another program wants to use Skype screen will come up. Enable Allow this program to use Skype, and click Ok to save.

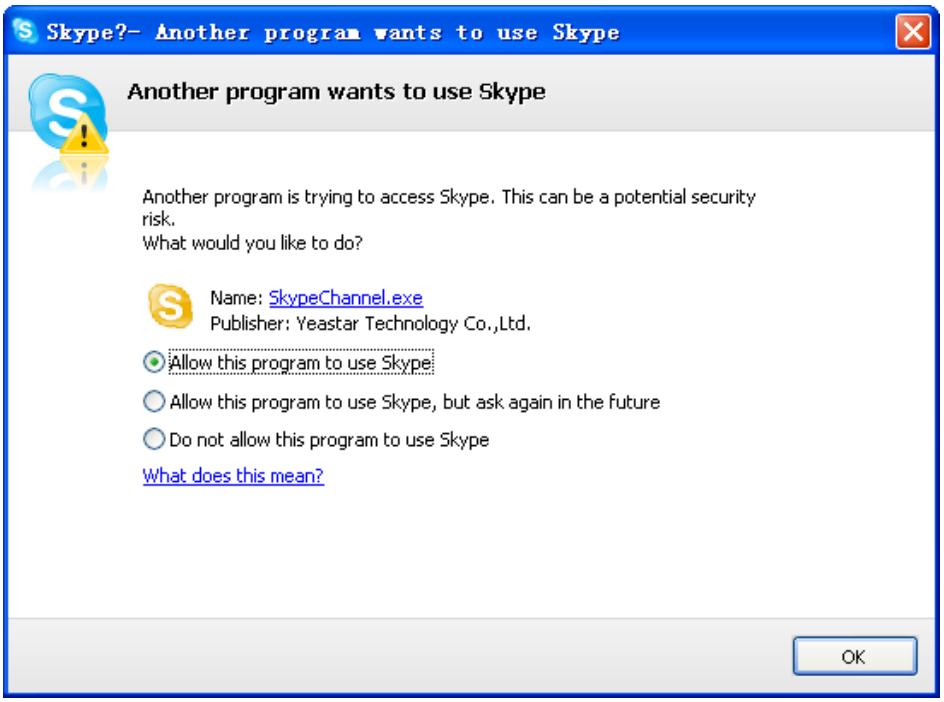


Figure 6-2

Step 3 Check if the status of all ports and Skype are absolutely normal

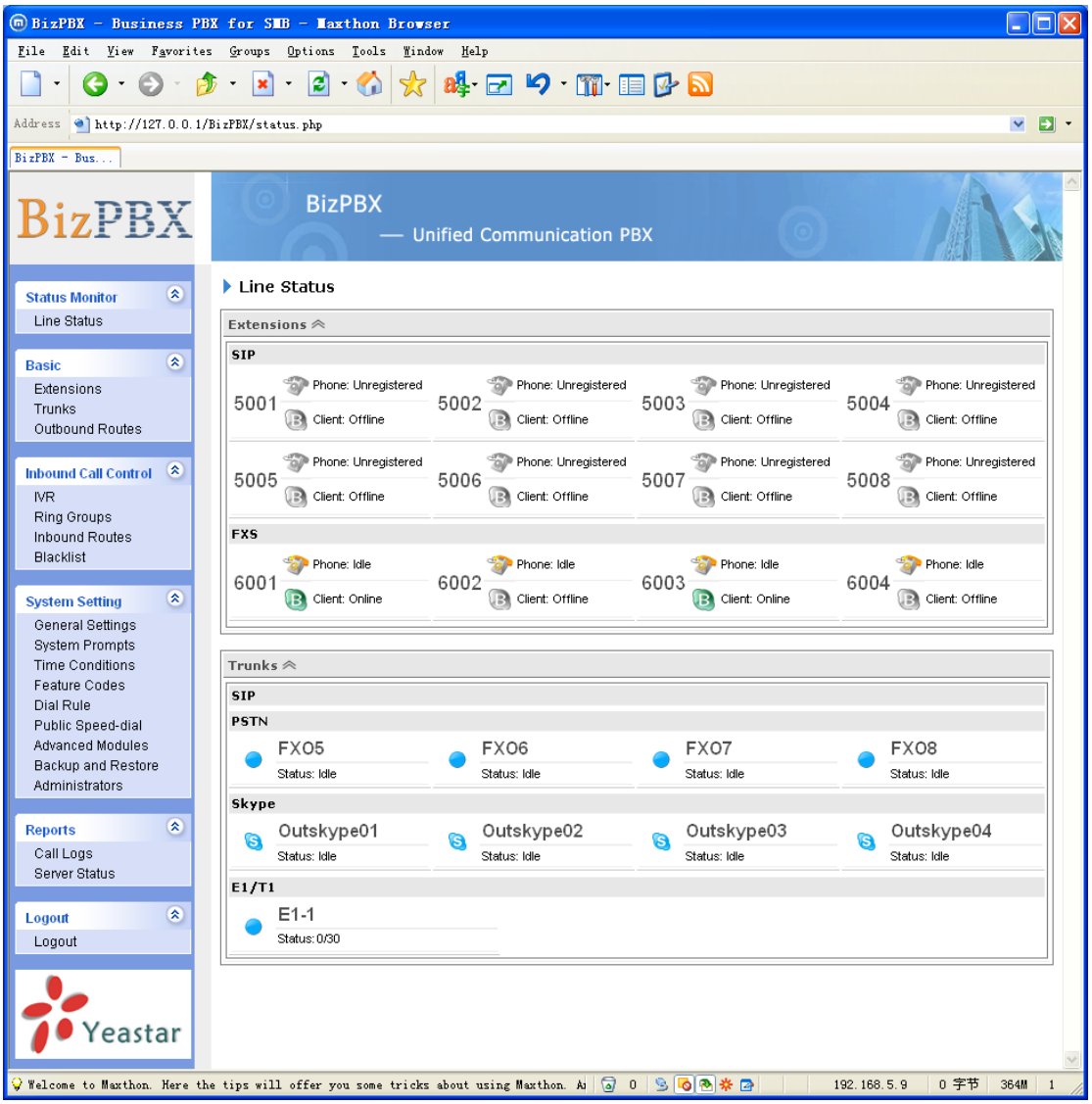

BizPBX - Business PBX for SMB - Maxthon Browser
File Edit View Favorites Groups Options Tools Window Help
Address http://127.0.0.1/BizPBX/status.php
BizPBX - Bus...
BizPBX
BizPBX
— Unified Communication PBX
Status Monitor
Line Status
Basic
Extensions
Trunks
Outbound Routes
Inbound Call Control
IVR
Ring Groups
Inbound Routes
Blacklist
System Setting
General Settings
System Prompts
Time Conditions
Feature Codes
Dial Rule
Public Speed-dial
Advanced Modules
Backup and Restore
Administrators
Reports
Call Logs
Server Status
Logout
Logout
Yeastar
Line Status
Extensions
SIP
5001 Phone: Unregistered Client: Offline
5002 Phone: Unregistered Client: Offline
5003 Phone: Unregistered Client: Offline
5004 Phone: Unregistered Client: Offline
5005 Phone: Unregistered Client: Offline
5006 Phone: Unregistered Client: Offline
5007 Phone: Unregistered Client: Offline
5008 Phone: Unregistered Client: Offline
FXS
6001 Phone: Idle Client: Online
6002 Phone: Idle Client: Offline
6003 Phone: Idle Client: Online
6004 Phone: Idle Client: Offline
Trunks
SIP
PSTN
FXO5 Status: Idle
FXO6 Status: Idle
FXO7 Status: Idle
FXO8 Status: Idle
Skype
Outskype01 Status: Idle
Outskype02 Status: Idle
Outskype03 Status: Idle
Outskype04 Status: Idle
E1/T1
E1-1 Status: 0/30
Welcome to Maxthon. Here the tips will offer you some tricks about using Maxthon.
192.168.5.9

Figure 6-3

7. Managing BizPBX Software

7.1 Open BizPBX Homepage

Double-click the shortcut 'BizPBX Management' on desk to run the software.

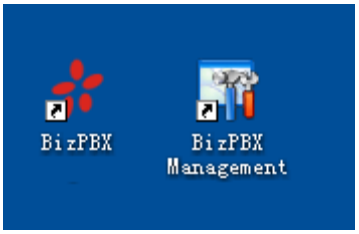


Figure 7-1

Or you can open the IE browser and enter in: http://127.0.0.1:8080

If you are not operating in BizPBX Server, please open the 8080 port of BizPBX Server through IE browser. For example, if the IP address of BizPBX Server is 192.168.0.101，then you can type in http://192.168.0.101:8080

7.2 Administrator Logging in

Open the BizPBX user login page. Input your username and password to gain access. Use the default administrator username 'admin' and the default password 'password' to login.

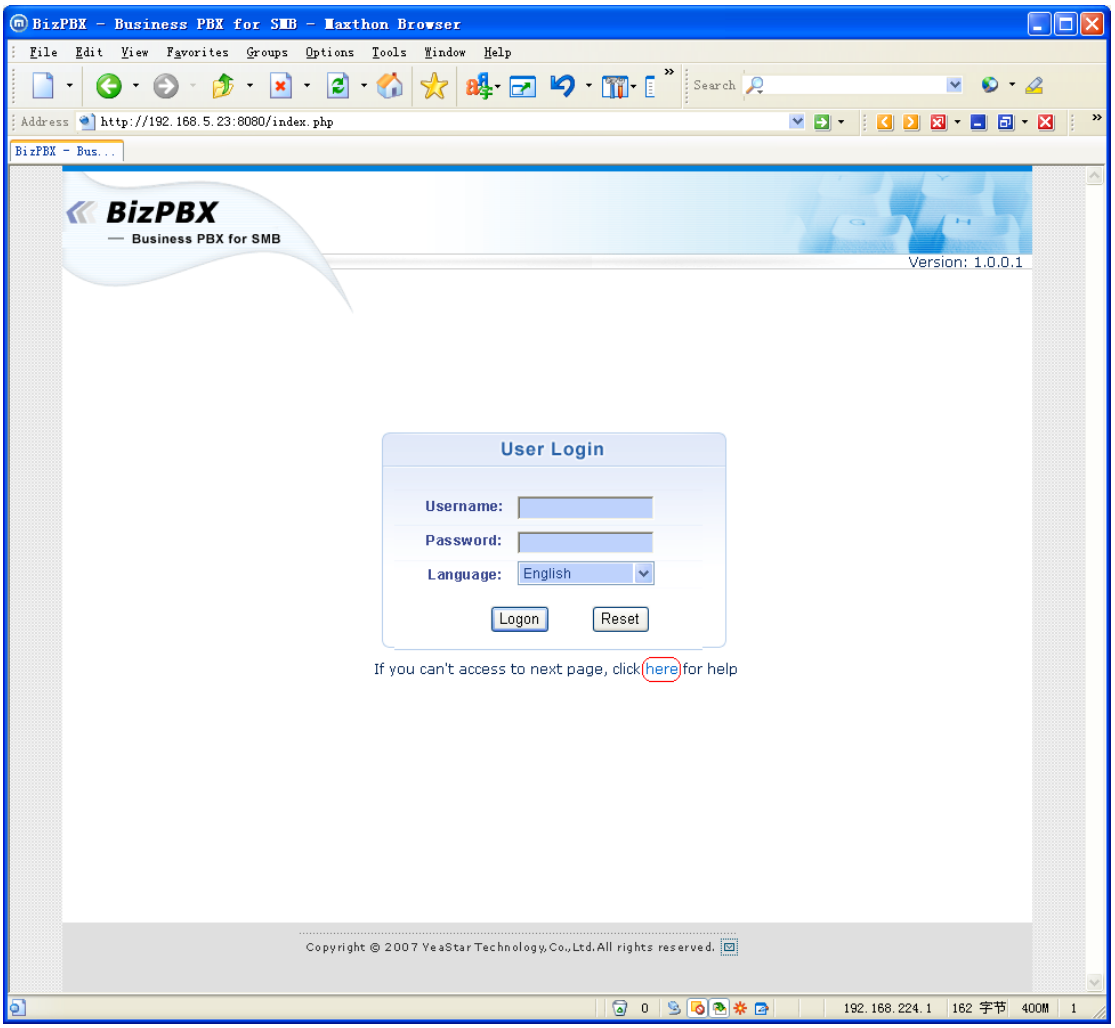


Figure 7-2

If the next page is abnormal after you login, that mainly because of the safety setting of IE browser. Click **here** link on this page and follow the guides to restore the normal page.

7.3 Status and Management

7.3.1 Line Status

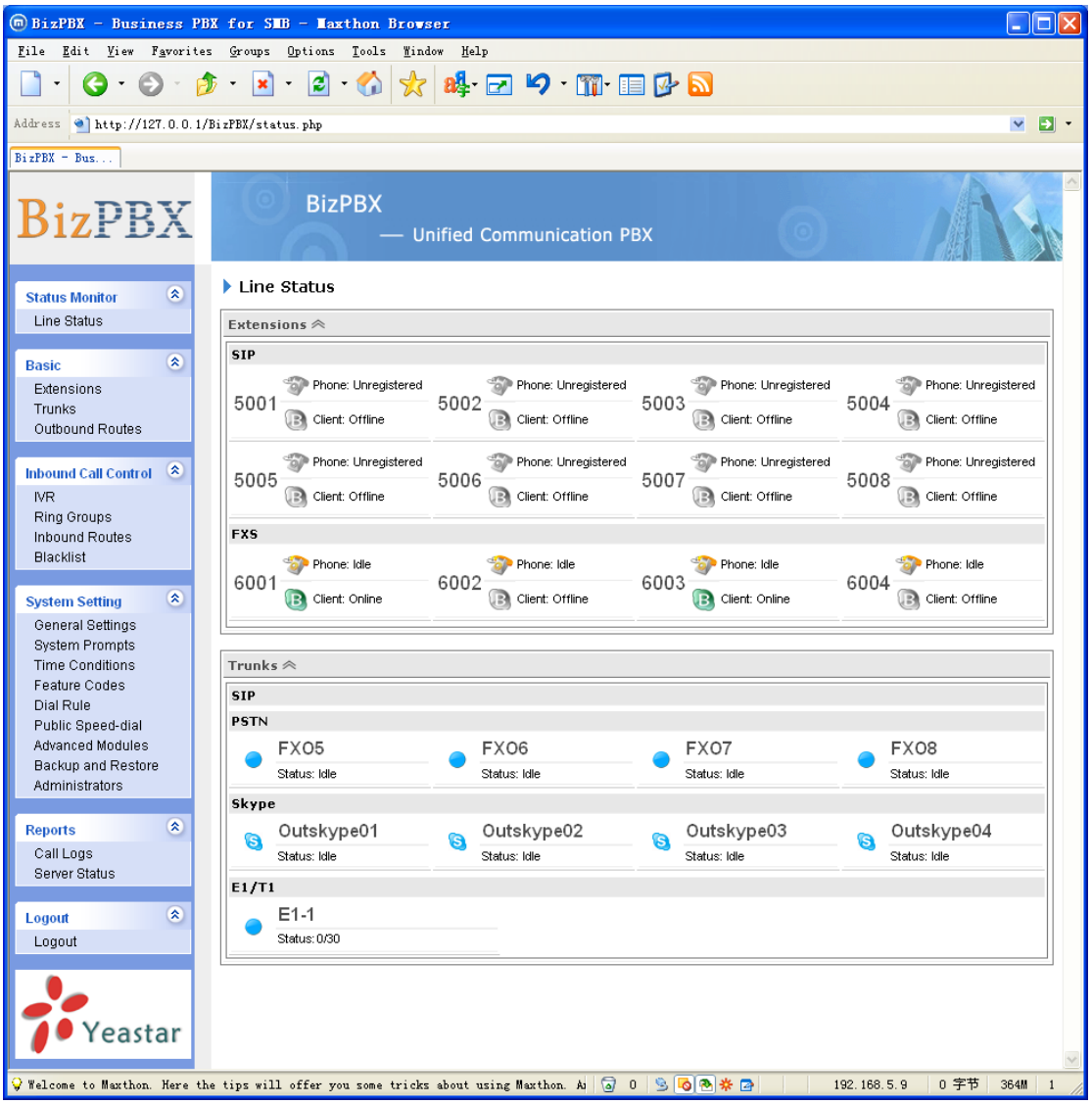


Figure 7-3

Description of all BizPBX items

Extensions:

SIP Extension:

SIP Phone Status:

1) : Phone is unregistered.

2) : Phone is idle.

3) : Phone is in service.

Client Status:

1） : Offline.

2） : Online.

FXS Extension:

Analog Phone Status

1) : Doesn't connect to analog phone/ Port is not available.

2) : Phone is idle.

3) : Phone is in service.

Client Status:

1） : Offline.

2） : Online.

Trunk:

SIP Trunk:

1) : Red— indicates failed initialization. Trunk registration failed.

2) : Blue— indicates succeeded initialization.Trunk is idle.

3) : Green—indicates succeeded initialization. Trunk is in service.

FXO Trunk:

1) : Red— indicates failed initialization.FXO port hasn't connected with PSTN line.

2) : Blue— indicates succeeded initialization.Trunk is idle.

3) : Green—indicates succeeded initialization. Trunk is in service.

Skype Trunk:

1) : indicates failed initialization. Skype is error.

2) : indicates succeeded initialization. Trunk is idle.

3) : indicates succeeded initialization. Trunk is in service.

E1/T1 Trunk:

1) : Red— indicates failed initialization.

2) : Blue— indicates succeeded initialization. Trunk is idle.

3) : Green—indicates succeeded initialization. Trunk is in service.

7.4 General Settings

7.4.1 Extension

Extension has two kinds: PSTN（FXS）and SIP Extension

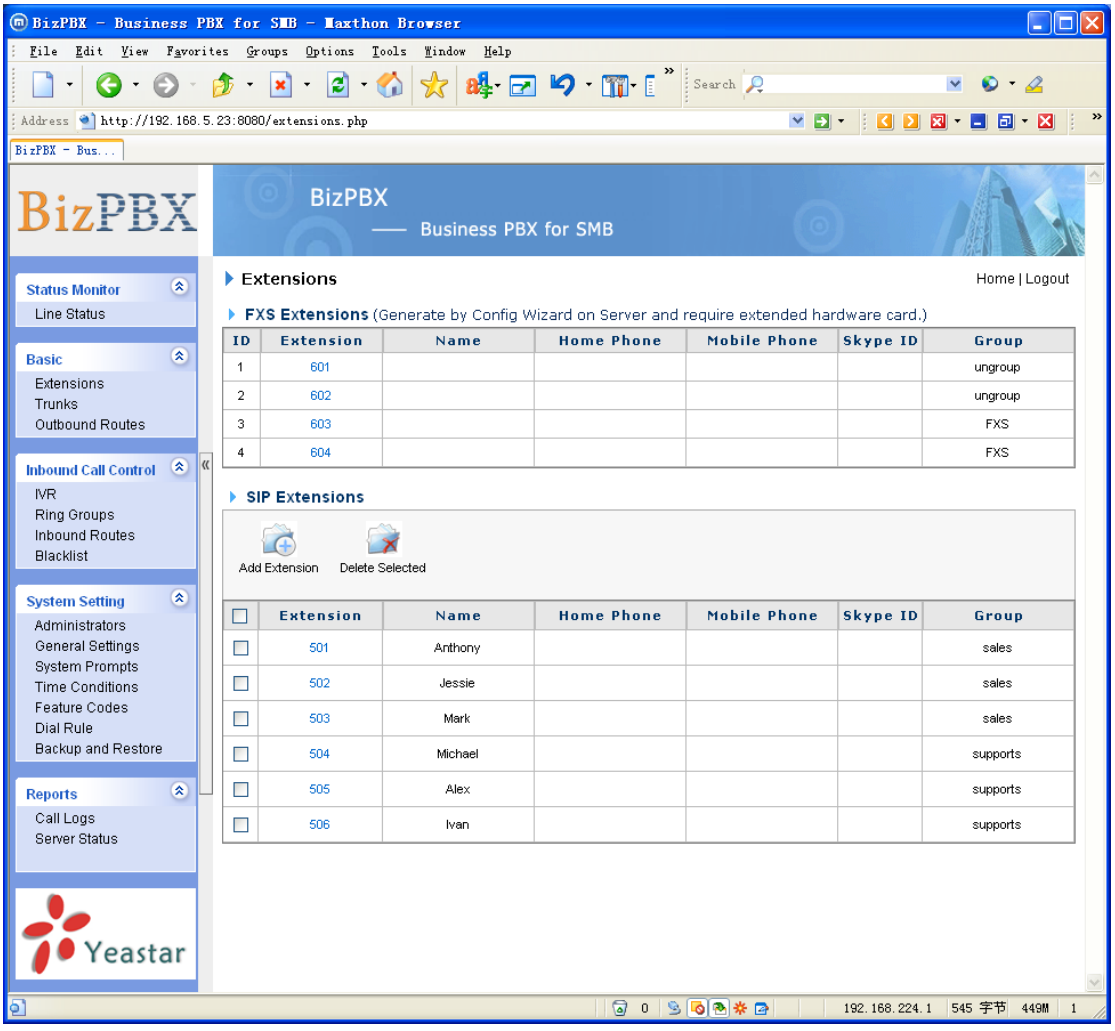


Figure 7-4

7.4.1.1 PSTN（FXS）Extension

1. Add PSTN（FXS）Extension

PSTN Exetension is configured through Serve's Config.Wizard, it needs the Yeastar's TDM800 hardware card support.

It's impossible to add PSTN Extension directly on homepage, but you can plug the TDM800 card with FXS modules, through this way you can get more PSTN extensions after finishing the 'Config.Wizard'. The newly generated PSTN (FXS) Extension will be showed as 'Undefined' and status is 'Error'; the extension can't be used until you configure it. Please refer to 'Edit PSTN (FXS) Extension'.

2. Edit PSTN（FXS）Extension

You should fill in the following blacks:

1. Extension Info
   - Account – Extension number
   - Password

   User can login the BizPBX Client on this account.

2. User Profile
   - Name – User Name/ Nickname
   - Group – group contacts on Client
   - Skype ID
   - Family Telephone Number
   - Mobile Phone Number
3. Call Forwarding

   The way of call forwarding. Here, administrator can set up the call transferring mode for extensions.
4. Administration Authority

   Admin can distribute the authority to extensions.

   Note: Here the authority means the administration right of BizPBX Client, such as surveillance, monitoring and automatic recording. When administrator distributes the right to an extension user and enables the advaced functions in 'Advanced Modules', this extension user will be able to monitor other extensions when logged on to Client.

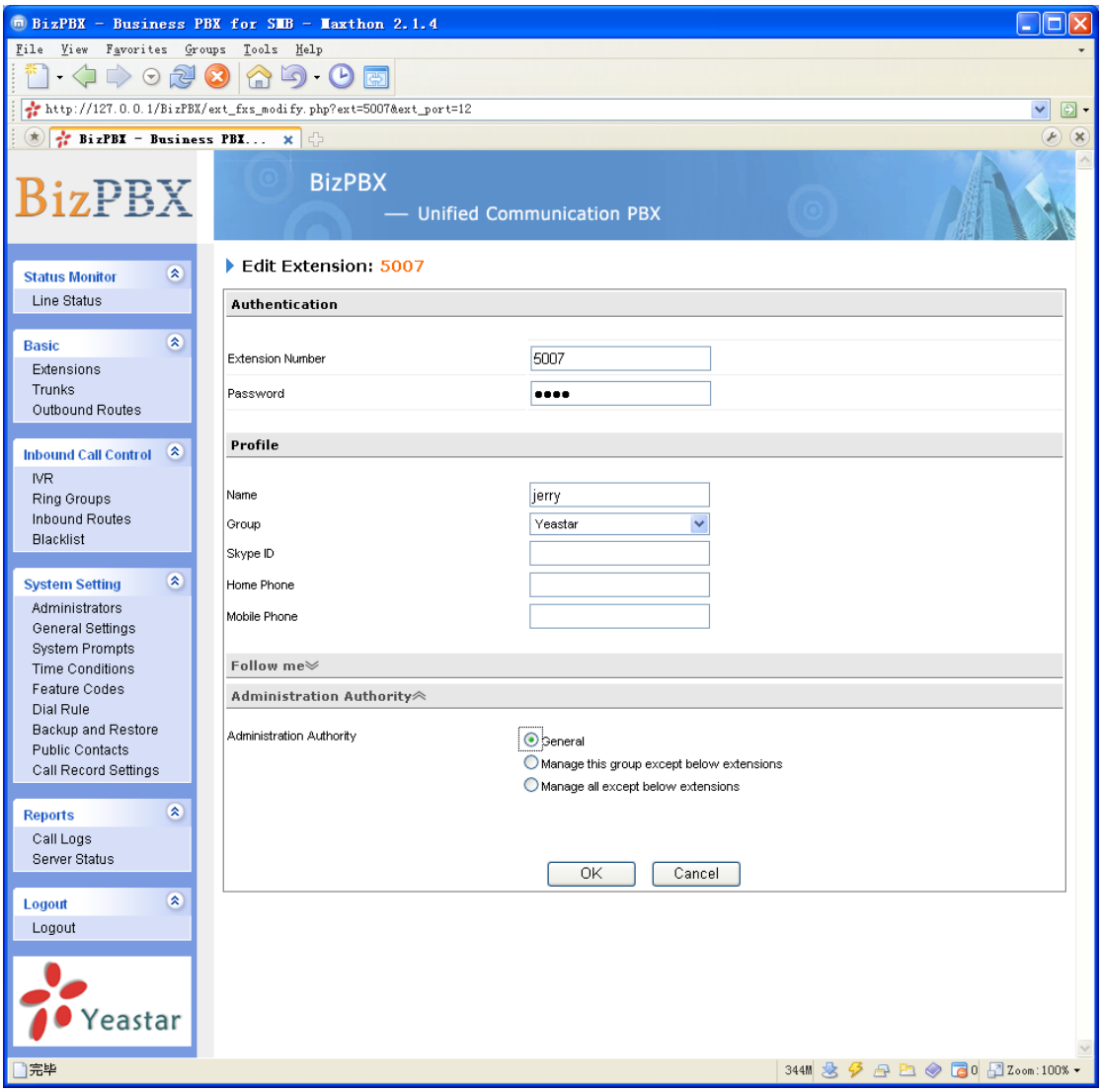


Figure 7-5

7.4.1.2 SIP Extension

1. Add SIP Extension

You should fill in the following blacks:

1. Extension Info
   - ·Account – Extension number （SIP number）
   - ·Password

   User can login the BizPBX Client on this account.

2. User Profile
   - ·Name – User Name/ Nickname
   - ·Group – group contacts on Client
   - ·Skype ID
   - ·Family Telephone Number
   - ·Mobile Phone Number
3. Call Forwarding

The way of call forwarding. Here, administrator can set up the call transferring mode for extensions.

4. Administration Authority

Admin can distribute the authority to extensions.

Note: Here the authority means the administration right of BizPBX Client, such as surveillance, monitoring and automatic recording. When administrator distributes the right to an extension user and enables the advaced functions in 'Advanced Modules', this extension user will be able to monitor other extensions when logged on to Client.

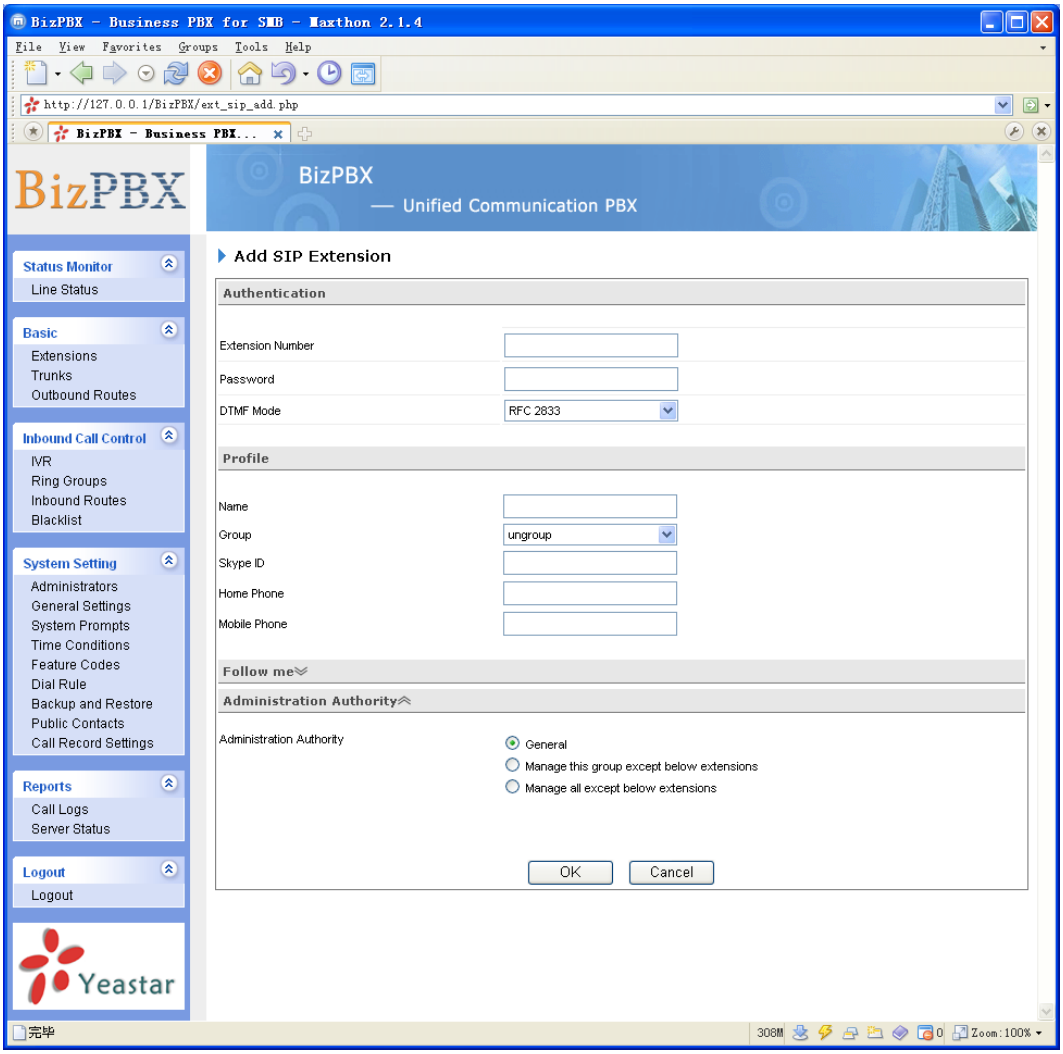


Figure 7-6

2. Remove SIP Extension

Click the extension that you want to remove, and click 'Delete Extension' key.

7.4.2 Trunk

It has three kinds of trunk, they are PSTN（FXO）,Skype and SIP trunks.

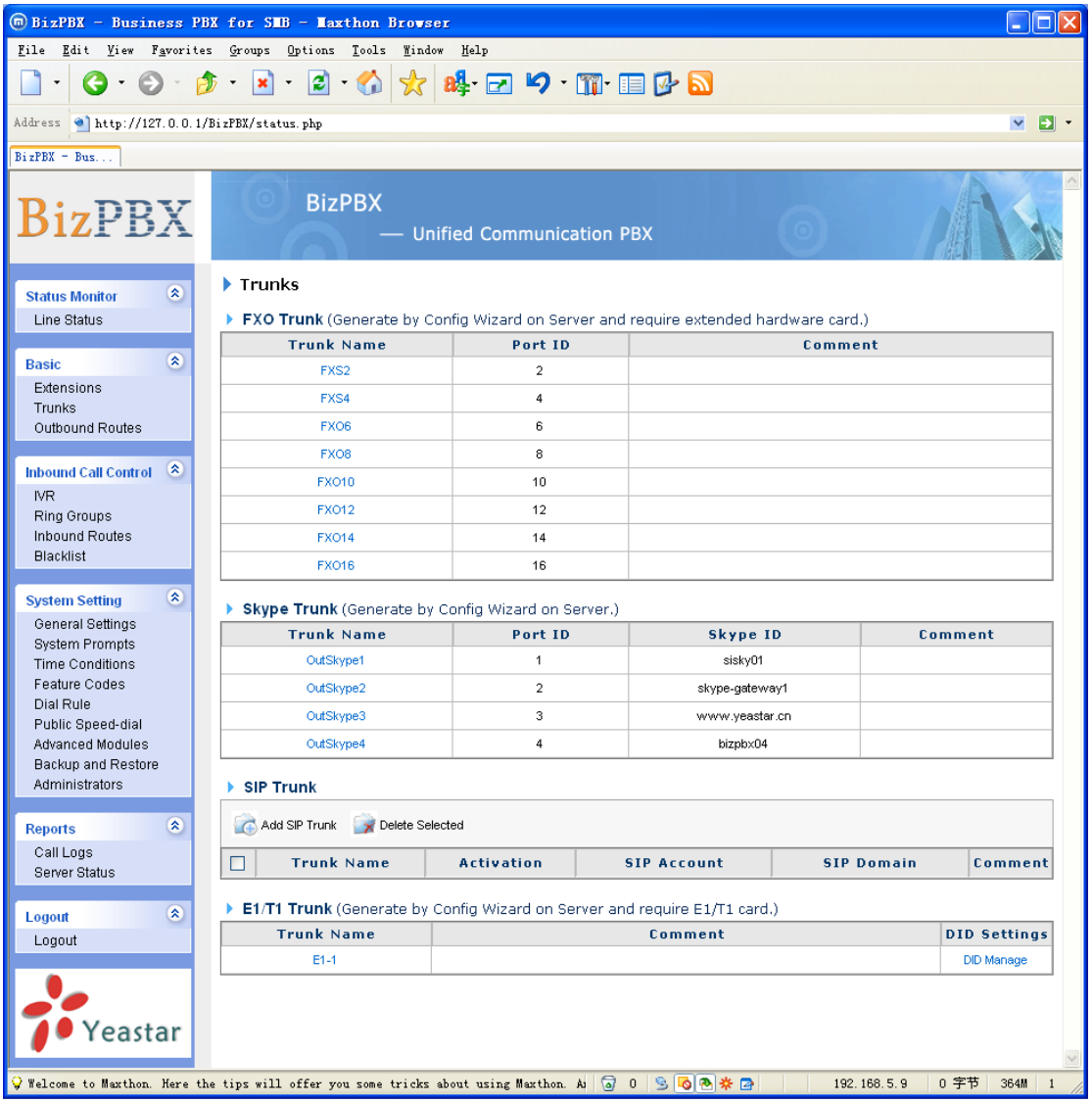


Figure 7-7

7.4.2.1 PSTN（FXO）Trunk

PSTN trunk is configured by Server Client's Config.Wizard, it needs the TDM800 hardware card support.

Add PSTN（FXO）Trunk

It's impossible to add PSTN trunk directly on homepage, but you can plug the TDM800 card with FXO modules, through this way you can get more PSTN trunks after finishing the 'Config.Wizard'. The trunk can be available when you plug PSTN line into the FXO port on the TDM800 card.

7.4.2.2 Skype Trunk

Skype trunk is configured by Server Client's Config.Wizard.

Add Skype Trunk

It's impossible to add Skype trunk directly on homepage, but you can get more Skype trunks through running 'Config Wizard'. Please refer to 5.2 and 5.3

• Trunk Name: the number connected with this trunk, just use as a marker.

·Activation: Whether or not enable this Skype trunk

·Skype Name: account of this Skype

·Skype Password: password of Skype account

·Busy Transfer: When the port is busy in service, it can transfer the following incoming calls to other idel trunks

1) If you enable 'No Transfer', system will hang up the call.

2) If you enable 'Transfer to Any idle Trunk', call will be transferred to next idle trunk.

3) If you enable 'To Customized Skype ID', the ID can be either a one in this server or not.

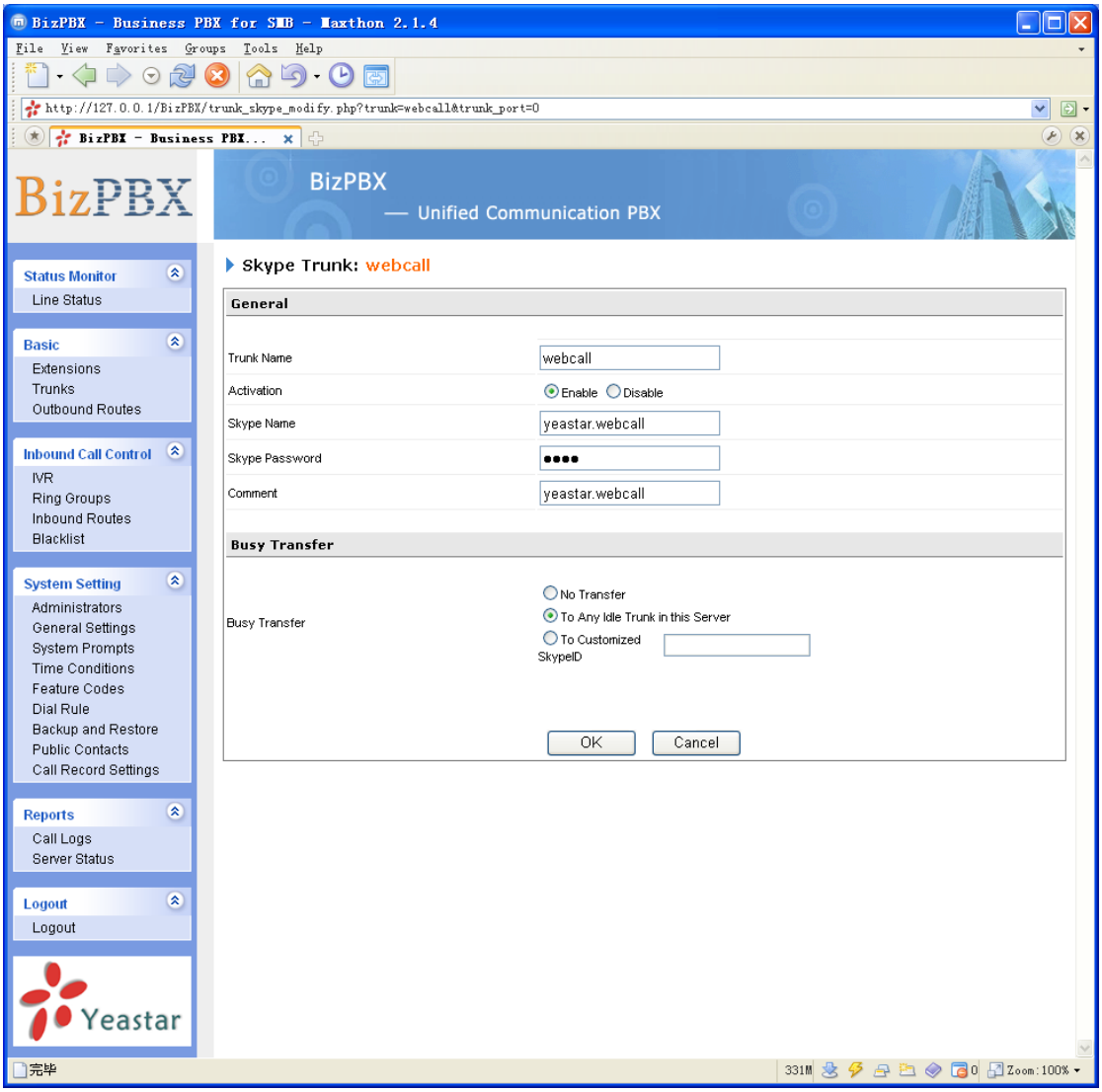


Figure 7-8

7.4.2.3 SIP Trunk

1. Add SIP Trunk

SIP trunk can not work until you fill in the SIP information correctly; you can ask VOIP Provider about details.

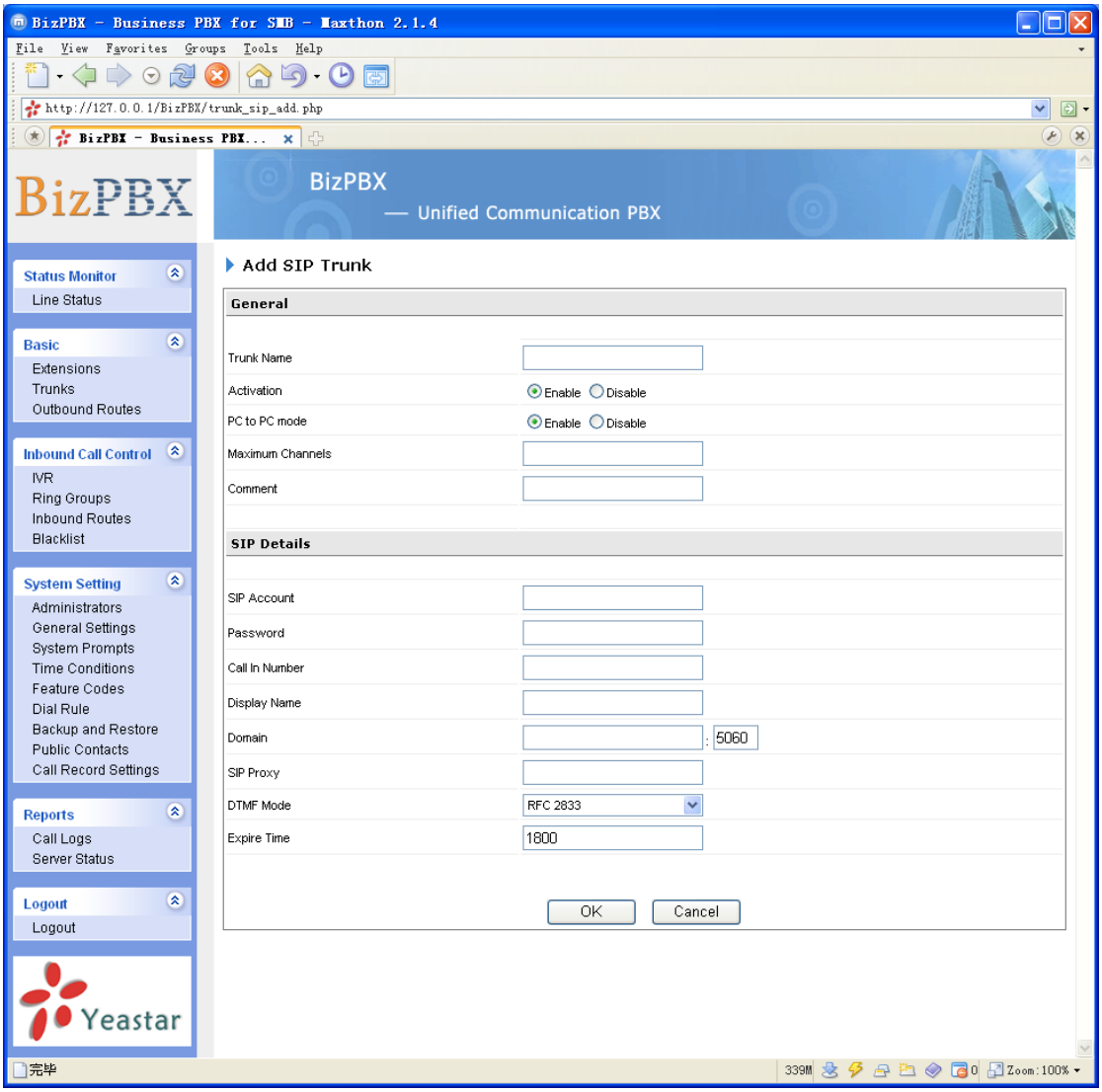


Figure 7-9

2. Remove SIP Trunk

Click the extension that you want to remove, and click ‘Delete Extension’ key.

7.4.2.4 E1/T1 Trunk

1. Add E1/T1 Trunk

Note: E1/T1 trunk is configured by Server Client’s Config Wizard, it needs the E1/T1 hardware card support.

It’s impossible to add E1/T1 trunk directly on homepage, but you can plug the YE110 card, through this way you can get the E1/T1 trunks after finishing the

'Config Wizard'.

2. E1/T1 Setting

Please see General Settings

3. Manage E1/T1 Trunk's DID

When configured E1/T1 on YE110 card, user is able to manage the DID of E1/T1 card. Please refer to the appendix E Manage E1/T1 DID

7.4.3 Outbound Routes

Outbound routing mainly works for gudies outgoing calls to go through trunks.

7.4.3.1. Buildup Outbound Routes

1）Click 'Add Outboud Route' and fill in the name of outbound route.

2）Dedicate routings for different type of calls.

On 'Apply this rule to' , please fill in the following blacks：

• Calls to Numbers starting with - To allow all those outgoing call numbers begin by this prefix number to use this outbound route.

For example, if set this prefix as 9, then all the telephone numbers that begin by 9 will call out through this routing.

• Calls from Extension(s)– Extensions allowed to use this routing.

• Auto attendant of outgoing calls – the auto attendant that allowed to use this routing. When someone makes incoming call to this auto attendant, he can dial telephone number to call out directly through this routing specially allowed auto attendant.

3）Appoint a calling way for outgoing calls

'Make outbound calls on'，you can choose the type of trunk

1、All trunks

2、All SIP trunks

3、All PSTN trunks

4、All Skype trunks

5、All E1/T1 trunks

6、Customized– User can appoint other type of trunk

4）Other

Deleted Prefix Count: If you set the number is 1, when you dial number 91234567, system will delete the first number 9, and the real dialing out number would be 1234567; if you set the number is 2, system will delete the 91, and the real dialing out number would be 234567.

Direct-Out number: If you set a number here, all the outging calls thourgh this route will guide to this number directly. It usually use for multi-branch connection.

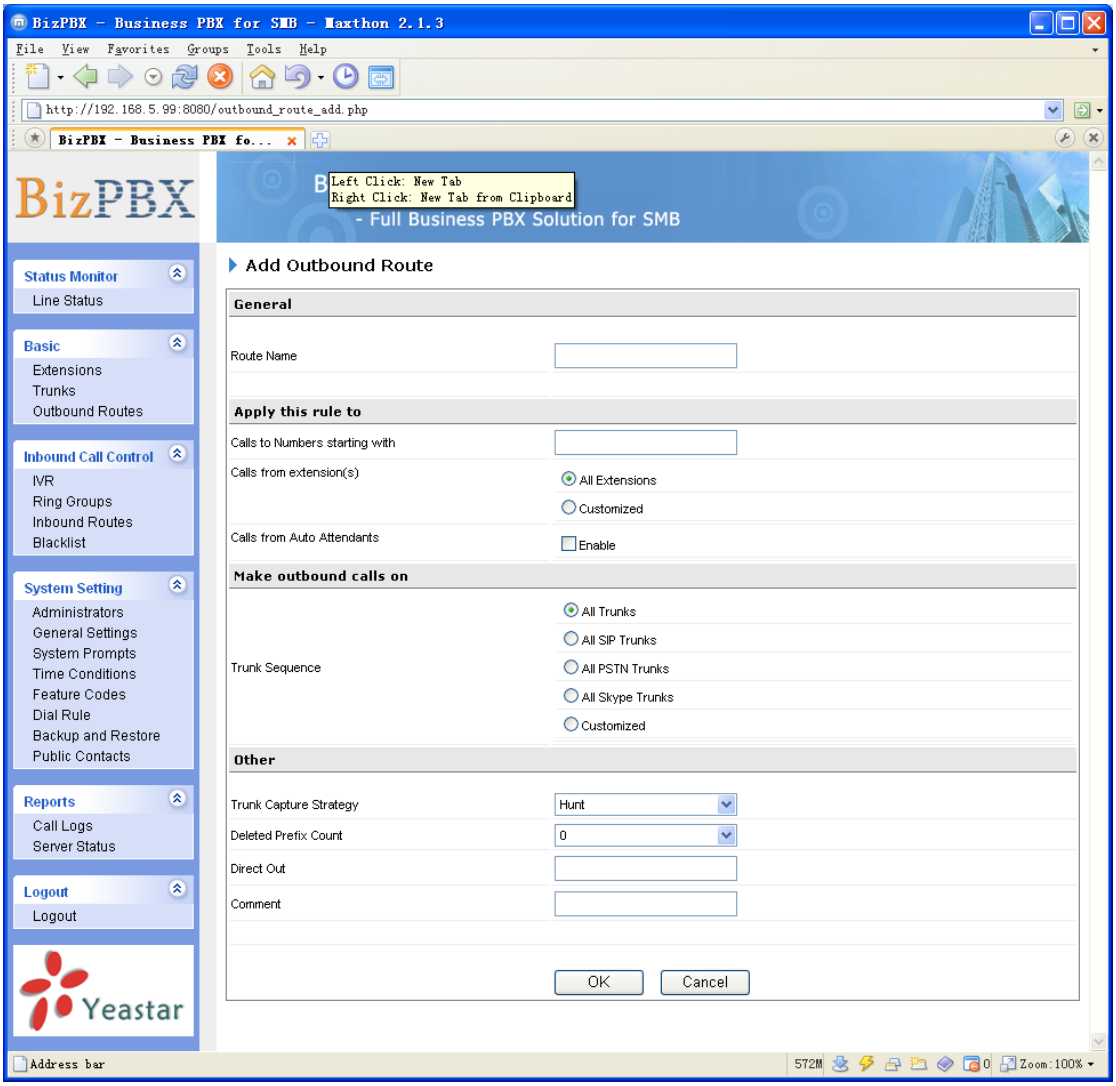


Figure 7-10

7.4.3.2. Remove Route

Click on the route and then click ‘Delete Selected route’.

7.5 Inboud Call Control

7.5.1 Auto Attendant

The auto attendant plays: Welcome to xxx Company. For product information, please press 1; For technical support, please press 2; For help press 0 or dial extension number directly.

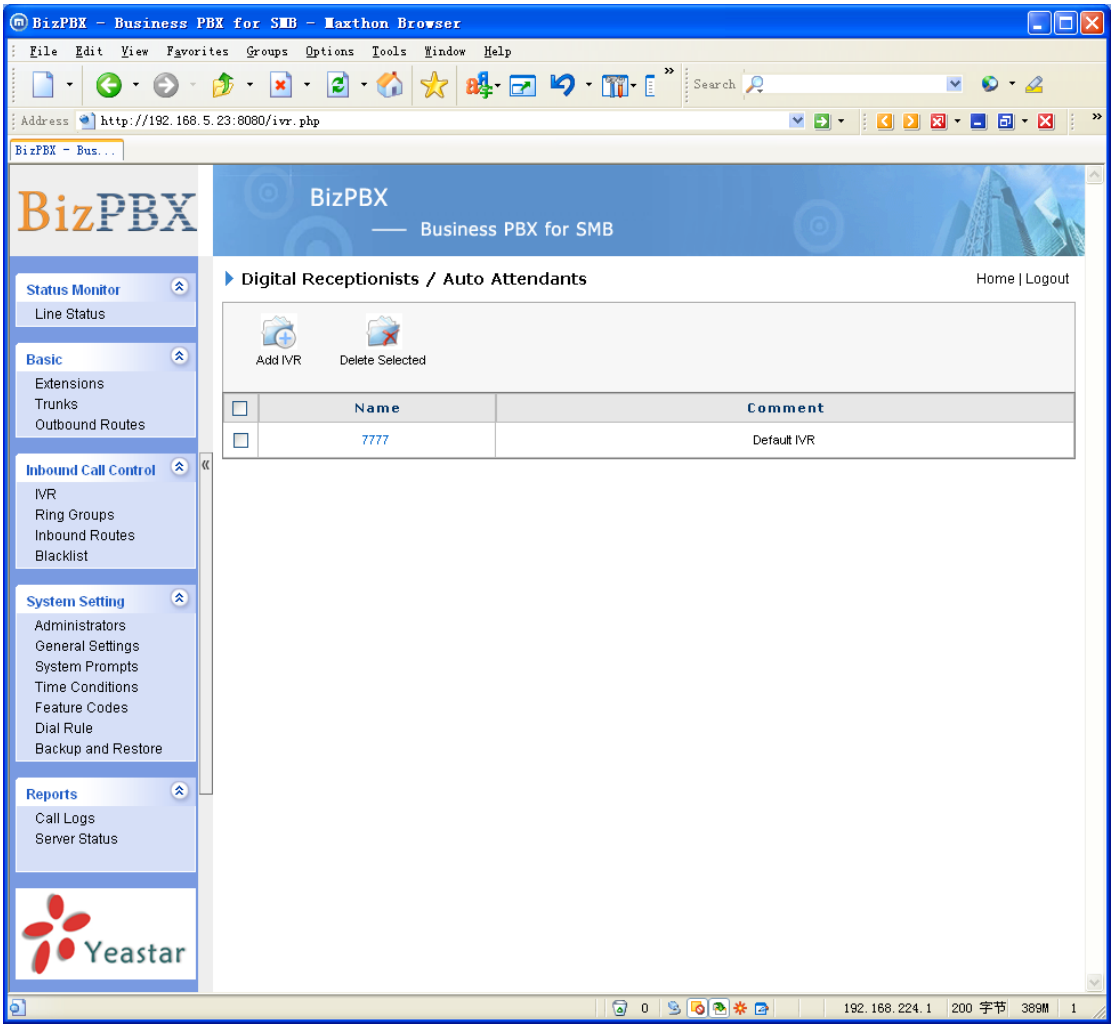


Figure 7-11

7.5.1.1 Setup Auto Attendant

Click 'Add Auto Attendant'

1）Name (Virtual Extension Number)

2）Click 'View' to choose the voice file you want to upload. This file will copy to BizPBX system \prompts\MusicOnHold catalog. Please refer to Appendix A

3）Whether to enable 'Direct Dial to Extension Number' or not

If disable it, user has no right to call extensions directly except press the number (DTMF) on IVR.

If enable it, user is able to call extension directly as well as pressing the keys provided by IVR (DTMF).

4）Enable or disable 'Direct dial to outline number'

If you diabled the function, when someone calls in and dial the out-line number directly after hearing the prompt sound, system will prompt call error.

If you enabled the function, when someone calls in and dial the out-line number directly after hearing the prompt sound, system will selete the corresponding outbound routing to call out.

Note: Out-line number refers the telephone number or mobile phone number.

5）Make IVR on Menu

Choose the action for corresponding key. You have three options here, Extension, IVR or Ring Group.

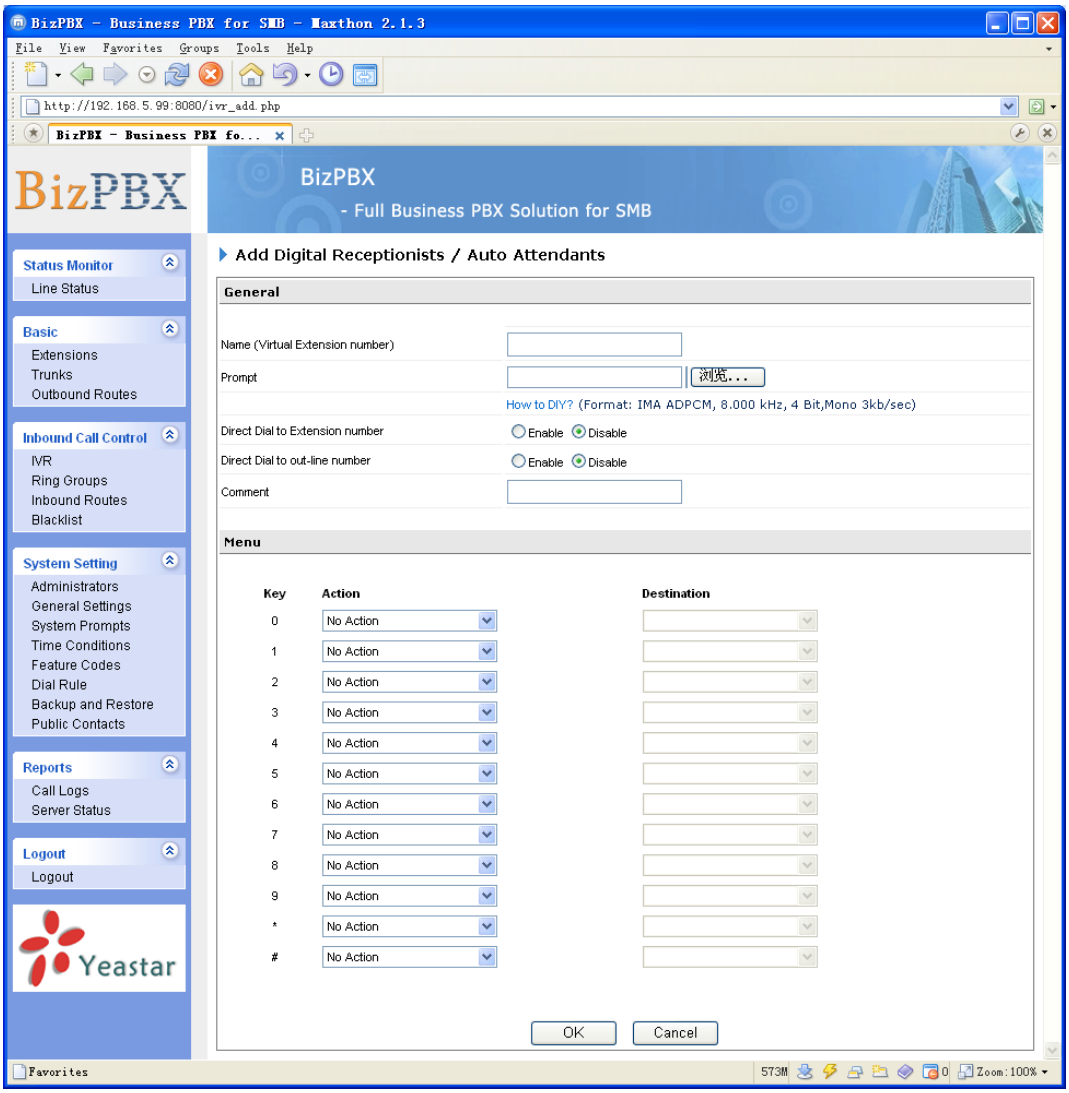


Figure 7-12

7.5.1.2 Remove IVR

Select the item you'd like to remove and click 'Delete Selected' key.

7.5.2 Ring Groups

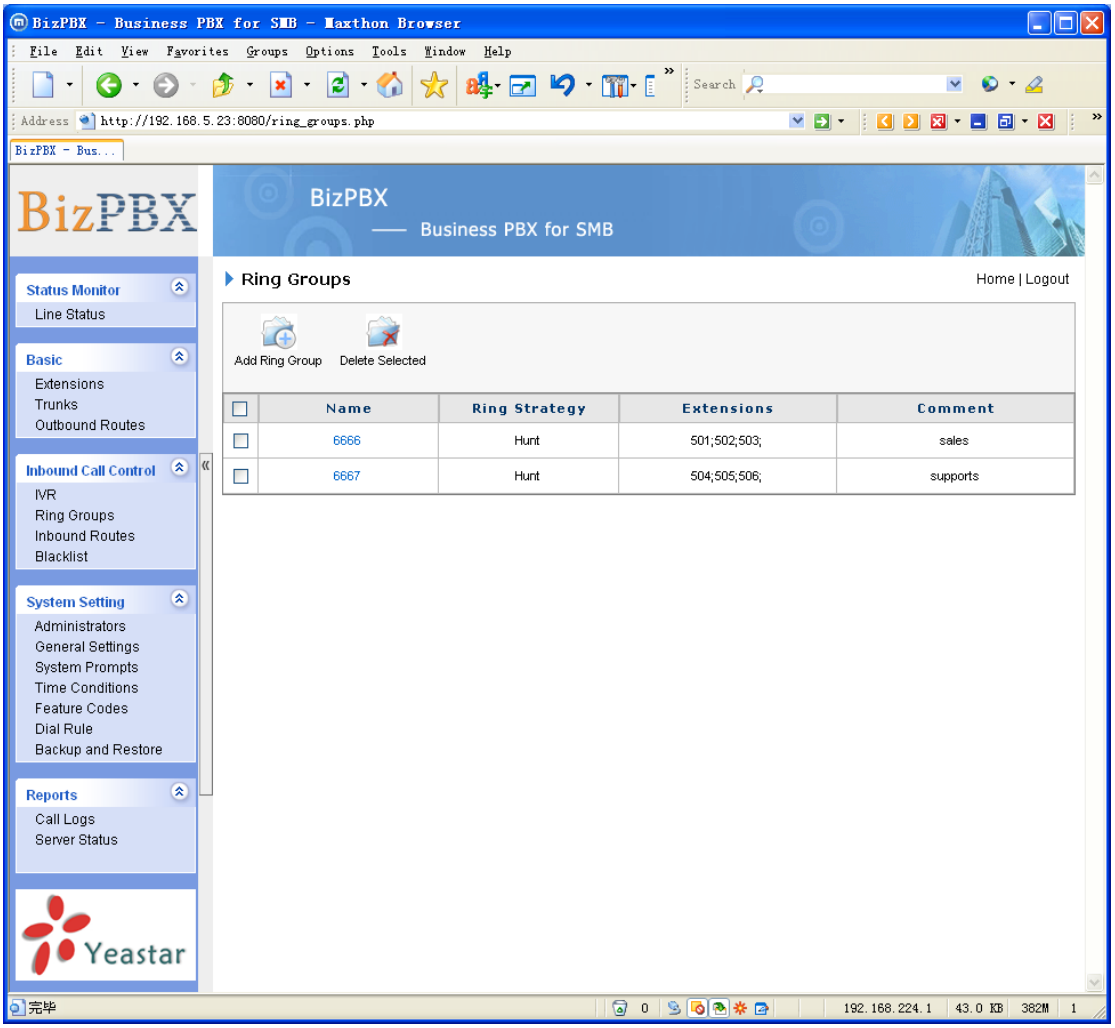


Figure 7-13

7.5.2.1 Create Ring Group

1）Click ‘Add Ring Group’

2）Input the required informations:

Group Name: It’s a virtual extension number that different from real extension number. Please distribute a number for this group.

Ring Strategy: You have two options.

·Hunt — it rings from the first extension to the next, one by one

·Ring all — all extensions on the group will ring at the same time

Ring time: the duration of ringing.

Music on hold: You can DIY the file. Please refer to Appendix A.

4）Ring Group Members

Click ‘>’ to move the selected extension into Members. Click ‘>>’ to move all extensions. You can have priority of them by Top, Up, Down or Bottom.

5）Destination of no answer

If none of the group members to answer the call, you can handle it by various ways, such as end the call directly, transfer to Extension, transfer to IVR or transfer to another group.

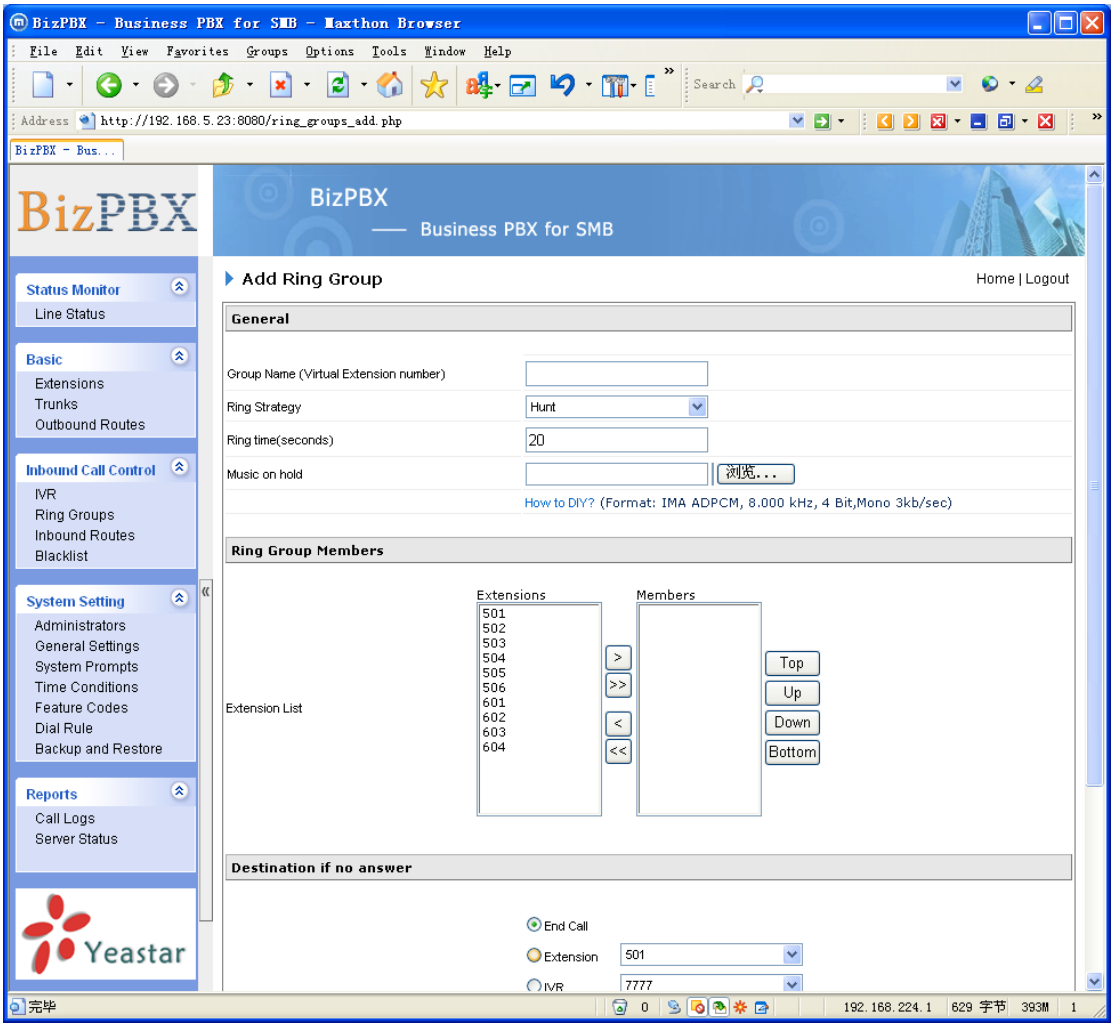


Figure 7-14

7.5.2.2 Remove Ring Group

Delete the selected group.

7.5.3 Queues

Call queues allow calls to be sequenced to one or more agents.

· **Queue Name** (Virtual Extension number): This option defines the numbered extension that may be dialed to reach this Queue.
· **Ring Strategy**: This option sets the Ringing Strategy for this Queue. The options are
1) RingAll: Ring All available Agents simultaneously until one answers.
2) Hunt: Take turns ringing each available Agent.
3) FewestCalls: Ring the Agent with the fewest completed calls.
4) Random: Ring a Random Agent.
5) RRmemory: RoundRobin with Memory, Remembers where it left off in the last ring pass.
6) ShortestTime: Ring the Agent which was least recently called
· **Max Wait Time**: The maximum number of seconds a caller can wait in a queue before being pulled out.
· **Music on hold**: Music (or Commercial) played to the caller while they wait in line for an available agent.

· **Queues Agents**: This selection shows all Users, Choose a User here makes them a member of the current Queue.

· **Destination if timeout**:
1) End Call
2) Extension
3) IVR
4) Ring Group
5) Queues

· **Announcements**
Is Announce:. Enabling this option causes BizPBX to report to the Agent, the hold time of the caller before the caller is connected to the Agent.

Frequency: How often to announce queue position and estimated holdtime (0 to Disable Announcements)
Custom Announcements Prompt:
For example: "Hi, you have entered the service queue, you are the n waiting person now." If the user wants to record this voice, as the number n is a random change (the system will automatically replace), so we have to split it into 3.
Voice 1: Type: Sound; File: upload a custom file (audio content: Hi, you have entered the service queue, you are); Order: 1
Voice 2: Type: Number; Order: 2

Voice 3: Type: Sound; file: upload a custom file (audio content: a wait person); Order: 3

· Other Options

Retry: The number of seconds we wait before trying all the phones again.

Wrap-Up-Time: How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 0, which is no delay.

Least Members: limit the number of members, when members of the queue is less than the limit value, the member will not quit.

IVR Break Out Menu: You can optionally present an existing IVR as a 'break out' menu. This is IVR must only contain single-digit 'dialed options'. The recording set for the IVR will be played.

Join Announcement: When members join the queue, this tone will be played.

Quit Announcement: When members quit the queue, this tone will be played.

Enable Custom Prompt: If enable custom prompt, it will play this prompt after the IVR break out menu prompt.

Note: Order of pleay queue prompt.

1) Play "Music on hold"

2) If Queue is busy, and enable Announcement, it will play "Custom Announcement Prompt".

3) And then it will play "IVR Break Out Menu" prompt.

4) If enable custom prompt, it will be played in the end.

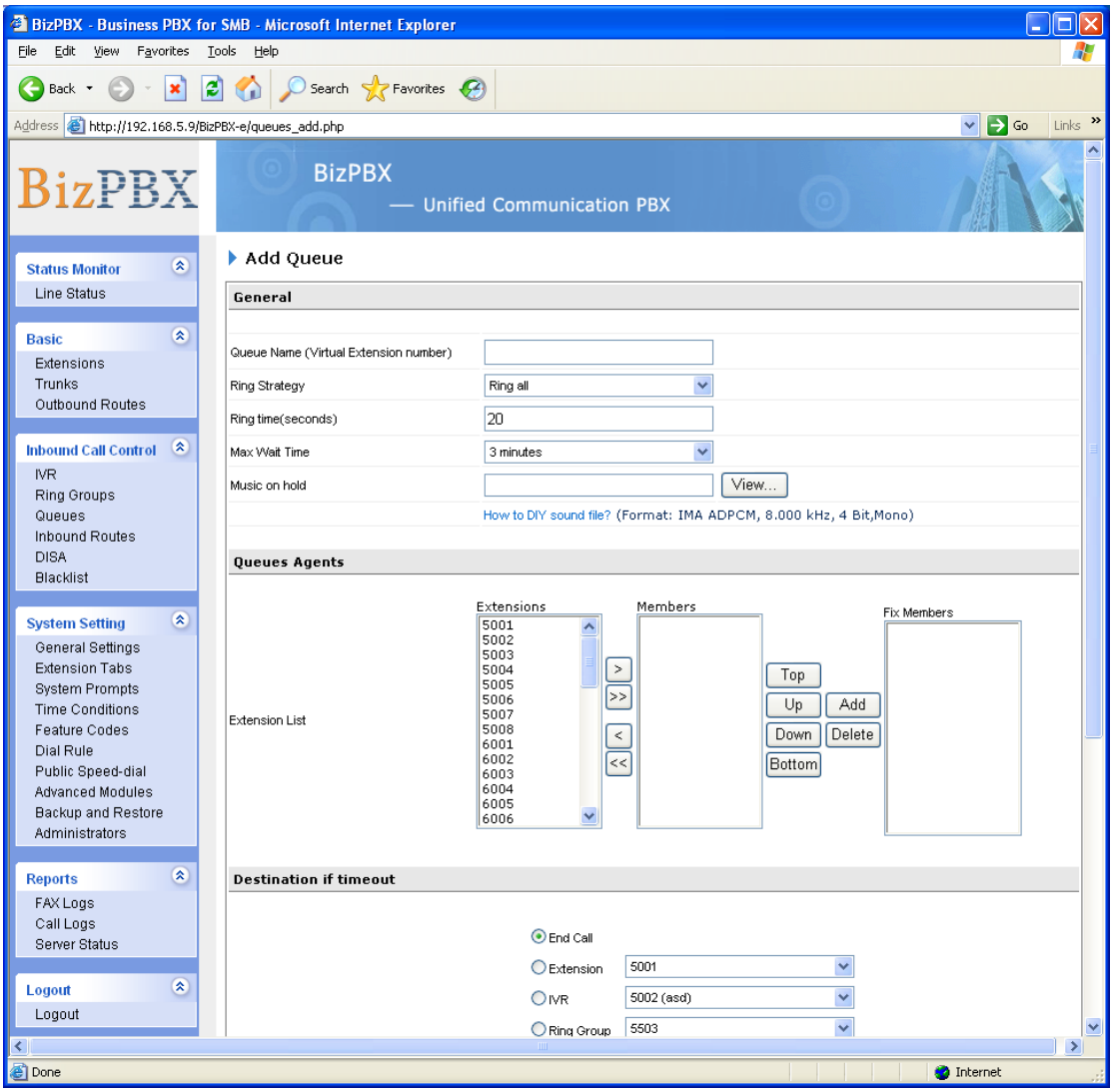


Figure 7-15

7.5.4 Inbound Routes

Inbound routing mainly works for gudies incoming calls to reach destination extension within the business hours or out of working hours.

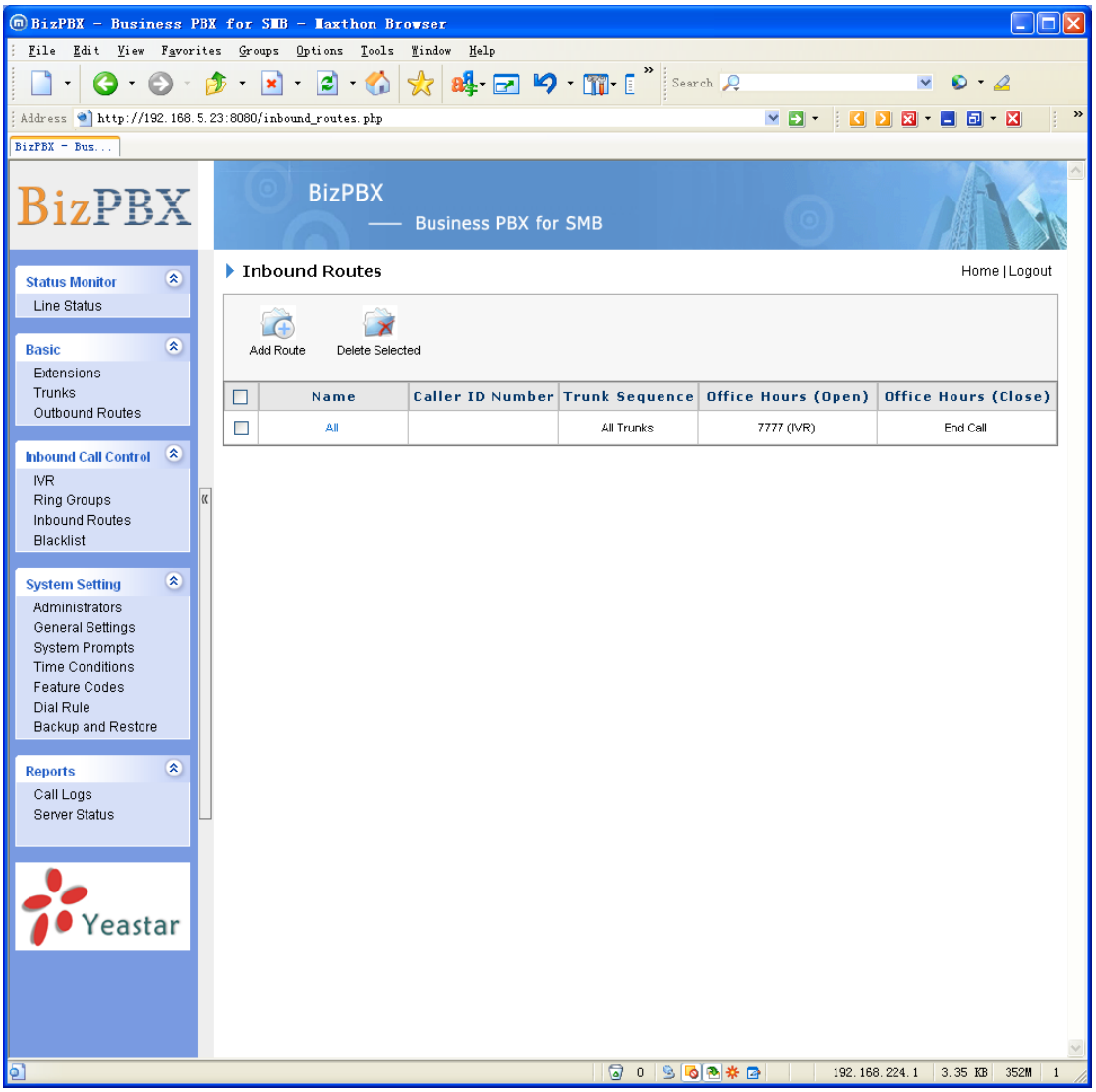


Figure 7-16

7.5.3.1 Buildup Inbound Routes

1）Click ‘Add Inboud Route’ and fill in the name of inbound route.

2）Dedicate routings for different type of calls.

Please input the number length on ‘Callin number length’

- Caller ID Number (DID/DDI number/mask)

Note: If empty caller ID number, it would be suitable for all incoming calls, and "00*" suitable for the calls that begin with 00.

You can enter multiple numbers here and separate them by semi-com.

3）Appoint the source for incoming calls

1、All trunks
2、All SIP trunks
3、All PSTN trunks
4、All Skype trunks
5、All E1/T1 trunks
6、Customized

4）Route calls during office hours to

You can choose the options for aswer incoming calls during office hours.

5）Route call outside office hours to

You can choose the options for aswer incoming calls during offwork time.

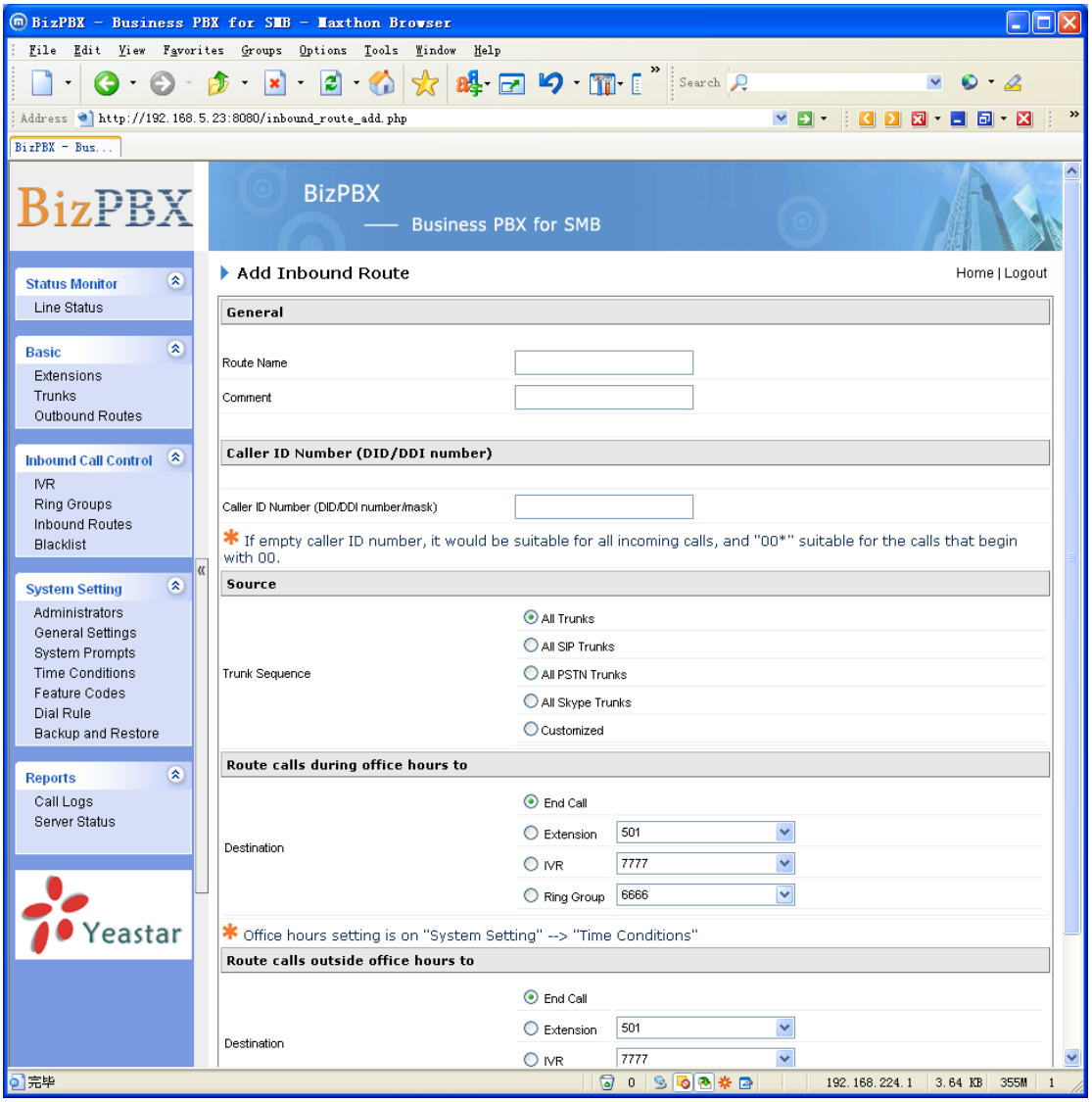


Figure 7-17

7.5.3.2 Remove Route

Click on the route and then click ‘Delete Selected route’.

7.5.4 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an ‘internal’ system dial tone and make calls as

if they were using one of the extensions attached to the telephone switch. To use DISA, a user calls a DISA number, which invokes the DISA application. The DISA application in turn requires the user to enter his extension number and extension password. If the extension number and password is correct, the user will hear dial tone on which a call may be placed. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security.

Note: if you enable the User's contact information verification, User use his contact information (mobile phone,home phone,skypeID) call in to the DISA, system will autoverify, if the information is correct, user will hear dial tone which a call may be palced.

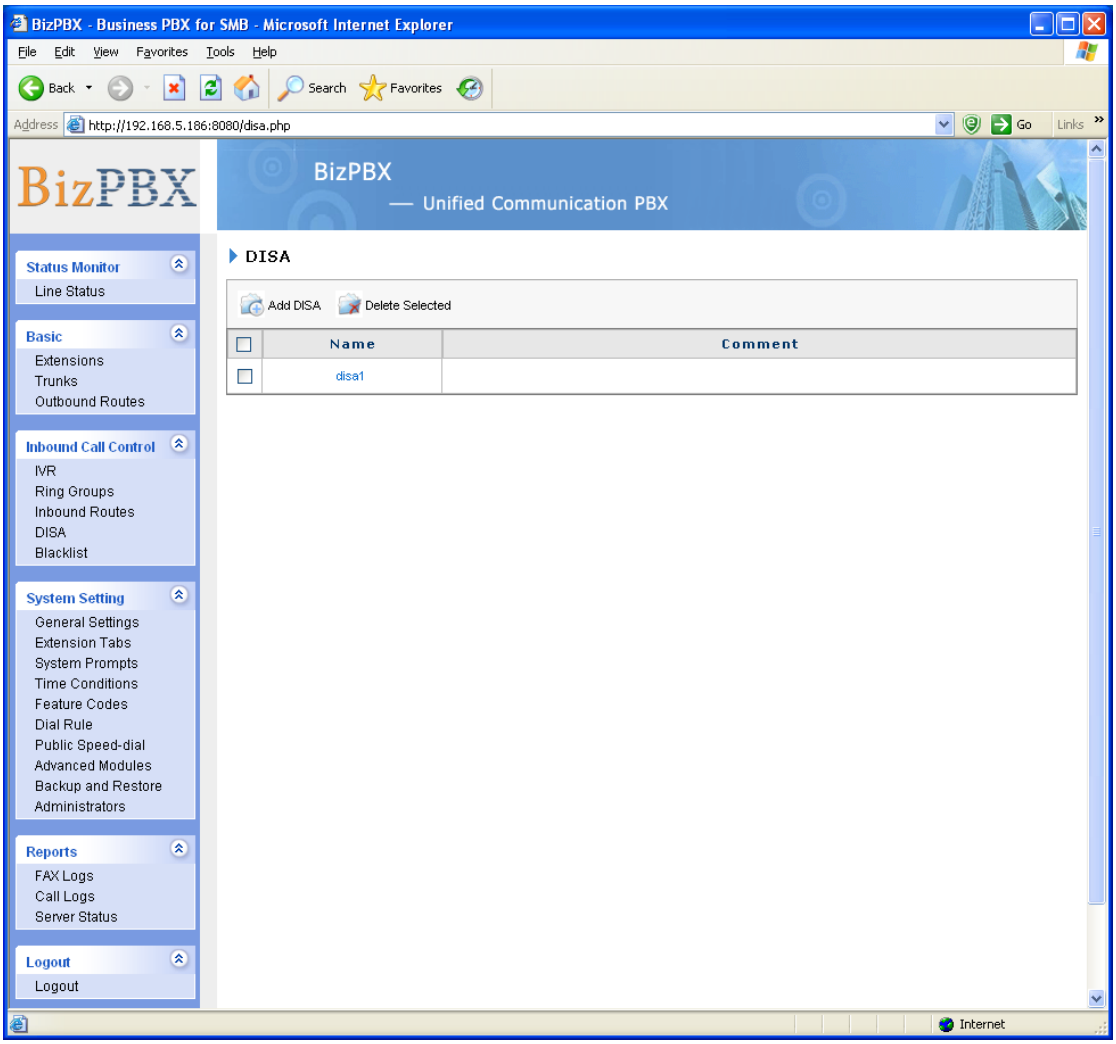


Figure 7-18

7.5.4.1 Create DISA

To add a new DISA application, click the New DISA button.

1) General

·Name

Give this DISA application a name to help you identify it.

·User's contact information (mobile, home phone, skypeID) verification

If enable the User's contact information verification, User use his contact information (mobile phone,home phone,skypeID) call in to the DISA, system will autoverify, if the information is correct, user will hear dial tone which a call may be palced.

·Comment

Give this DISA a comment to help you identify it.

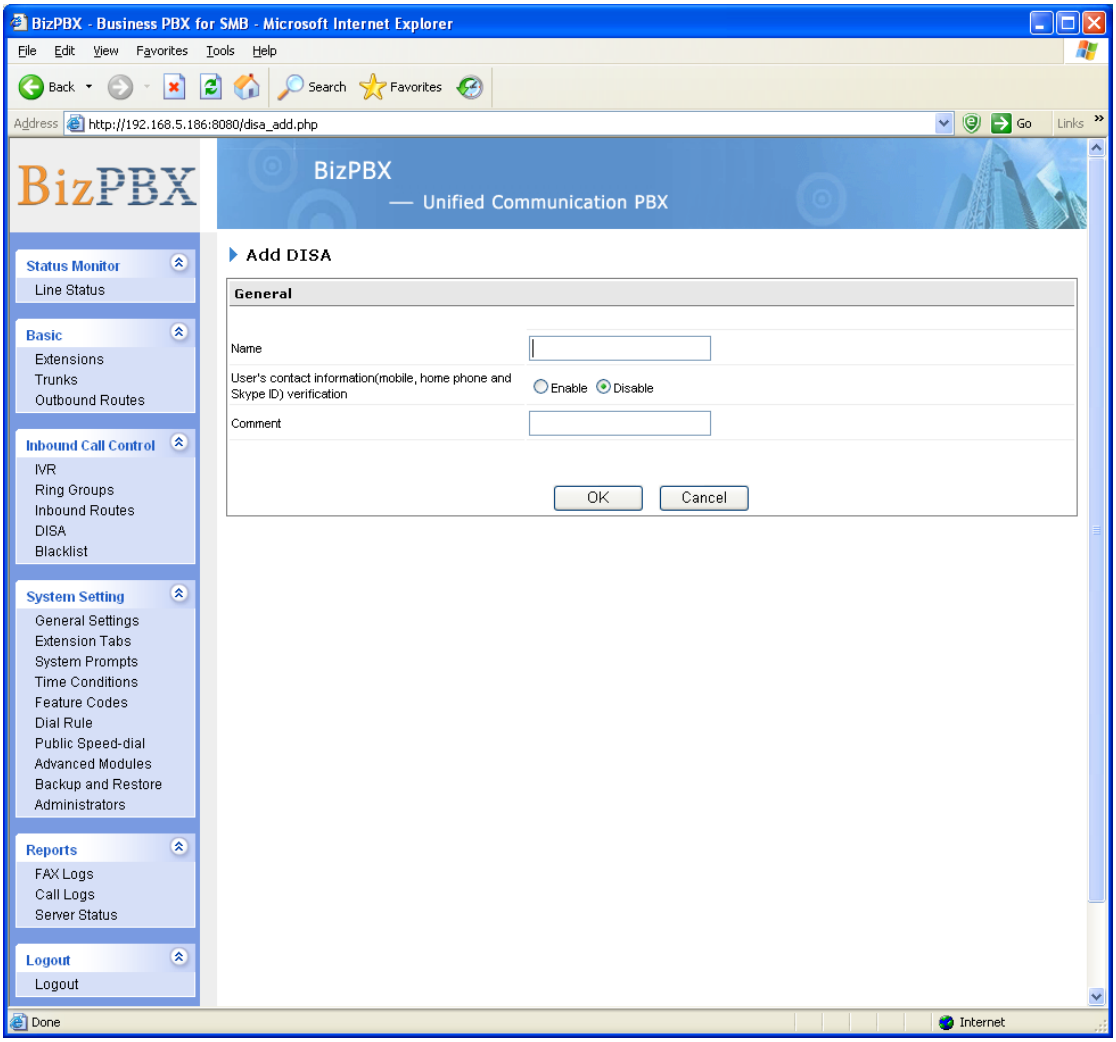


Figure 7-19

7.5.4.2 Remove DISA

Click on the DISA and then click 'Delete Selected'.

7.5.5 Blacklist

Blacklist can forbid bootie calls.

7.5.5.1 Add Blacklist

1）Click ‘Add Blacklist’ .

2）Input the annoying number or SkypeID on the item ‘Number’ to forbid all the incoming calls from this number or ID.

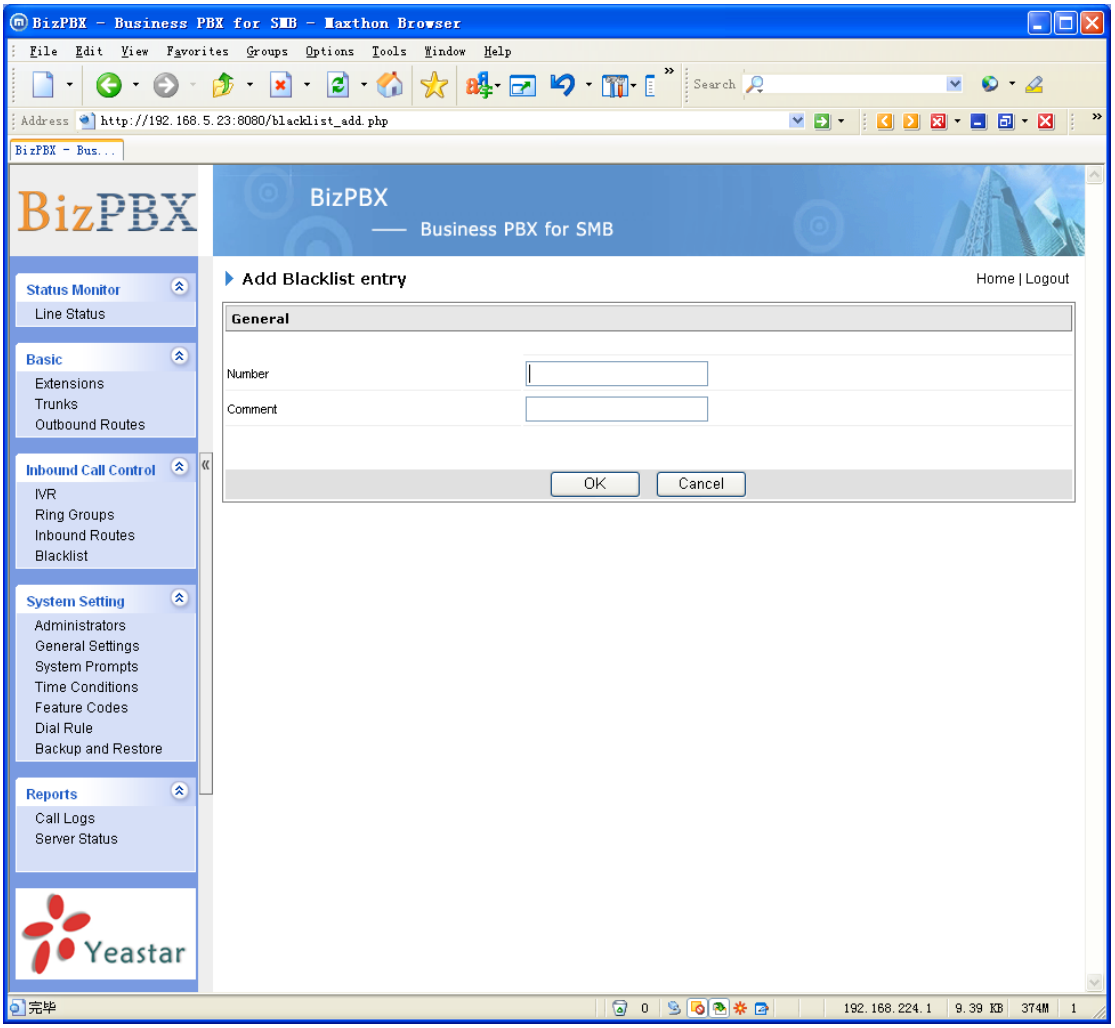


Figure 7-20

7.5.5.2 Remove Blacklist

Selected the item and then click ‘Delete Blacklist’ key.

7.6 System Settings

7.6.1 Gnereal Settings

1. Country
   Select the country to match with the country's tone cadence.
2. Port Configuration
   SIP port and RTP port are dedicated for SIP usage, please don't modify them.
3. Codec
   The default codecs are G.711 U-Law Codec，G.711 A-Law Codec and GSM 6.10 Codec. BizPBX also compatible with G.729 Codec and G.7231 Codec
   Notes: please make sure you have got the license of G.729 and G.7231 before using them.
4. STUN Server Settings
   Enable STUN Server for extensions.
5. Extend Card Settings
   1) Caller ID Type
   This option allows you to define the start of a Caller ID signal:
   After Ring: Start after a ring is received.
   Before Ring: Start before a ring is received.
   NTT-Japan: this is for Japan, Start when a polarity reversal is started.
   2) Speaker Volume: set the speaker volume for the extend card.
   3) Microphone Volume: set the microphone volume for the extend card.
   4) NTT-Japan

6. E1/T1 Settings
   1) E1/T1 Switch Type: Unknown,NI-1、NI-2,DMS 100,Lucent 5E, AT&T 4ESS,EuroISDN E1,EuroISDN T1,GR-303 EOC,GR-303 TMC,QSIG.
   2) E1/T1 Node Type: pri_cpe or pri_net
   3) E1/T1 DTMF Type: inband or pri_info
   4) E1/T1 Codec Type: A_Law or U_Law
   5) E1/T1 CallerID: when extension make call via E1/T1 Trunk, callee will display this caller id.
7. Timeout Settings
   1）Waiting time to dial first number（DTMF）
   2）Interval time on dailing（DTMF）：
   3）Incoming call timeout：
   4）Outgoing call timeout：
   5）Call on hold timeout：
   6）Voicemail timeout：When the max. voicemail duration exceeded, call will be hang up. 。

8. Flash Time Setting

Flash Time

9. File saves Path

1）Voicemail save path

If you empty it, file will be saved on *sounddata* folder under program path

2）E-Mail attachment save path

If you empty it, file will be saved on *attachments* folder under program path

Note：Save path must be on Server port. (format：C:\BizPBX)

10. Others:

1）Keep call history

2）keep voicemail for

3）Keep recordings for

4）Keep auto-records for

You have options 'Not Save', 'Save a moth', 'Save 3 months', 'Save 6 months and 'Save forever'

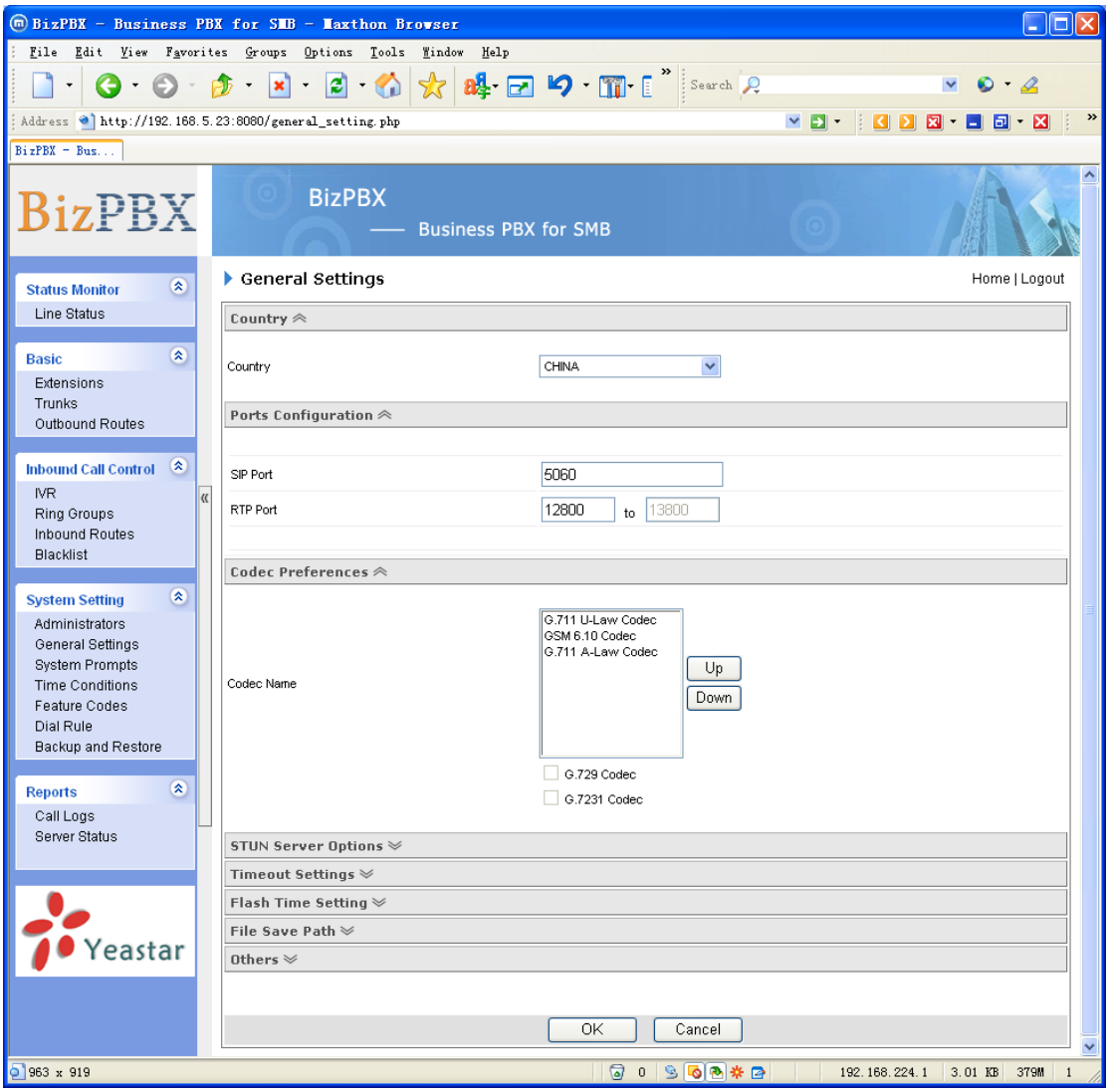


Figure 7-21

7.6.2 System Prompts

1）Dial Tone
2）Ring Back Tone: Caller hears this sound after get through.
3）Music on Hold: Caller hears this music when the call is hold.
4）Transfer to voicemail: Prompt the caller that the call will be transferred to voicemail.
5）Call Transfer failed: Prompt the caller that the call transferred failed.
6）Call failed: Prompt the caller that the call failed.
7）Call Busy: Prompt the caller that callee is busy now.
8）Invalid key in IVR: Prompt the caller that the key pressed is invalid.
9）Finish listening voicemail: Play this sound to prompt caller when over listening voicemail
10）Skype Extension incoming call
When there's a incoming call come from the Skype Extension, BizPBX will play this Prompt and guide the caller to make call.
11) Extension no answer.
12) Prompt sound for entering extension number after DISA.
13) Prompt sound for entering authentication code after DISA.
14) Prompt sound for authentication error after DISA.
15) The dial tone that will be played after entered DISA.
16) Prompt sound for dial error after entered DISA.
17) A total number of voice mails.
18) Number of new voice mail.

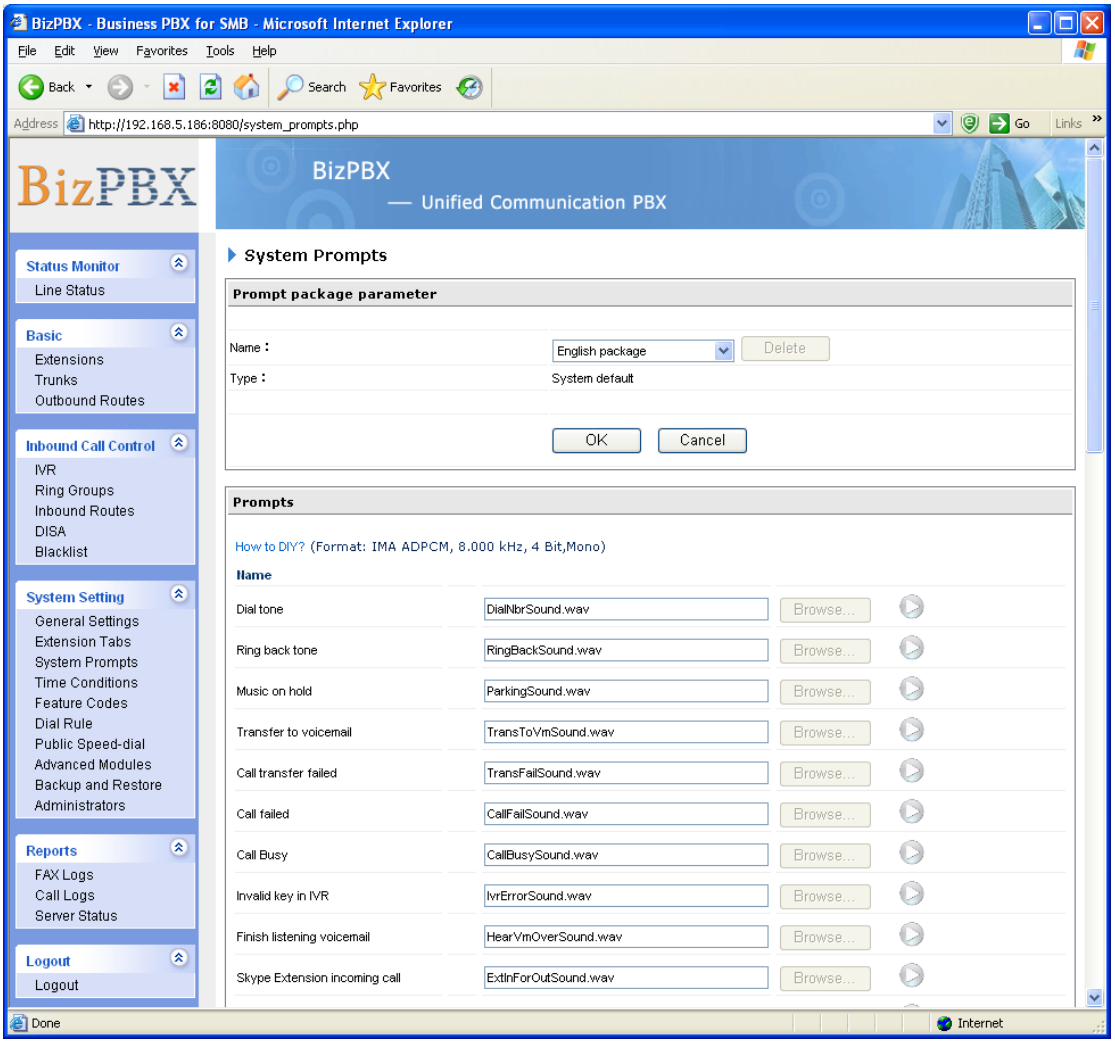


Figure 7-22

7.6.3 Time Conditions

You can setup the business hours here.

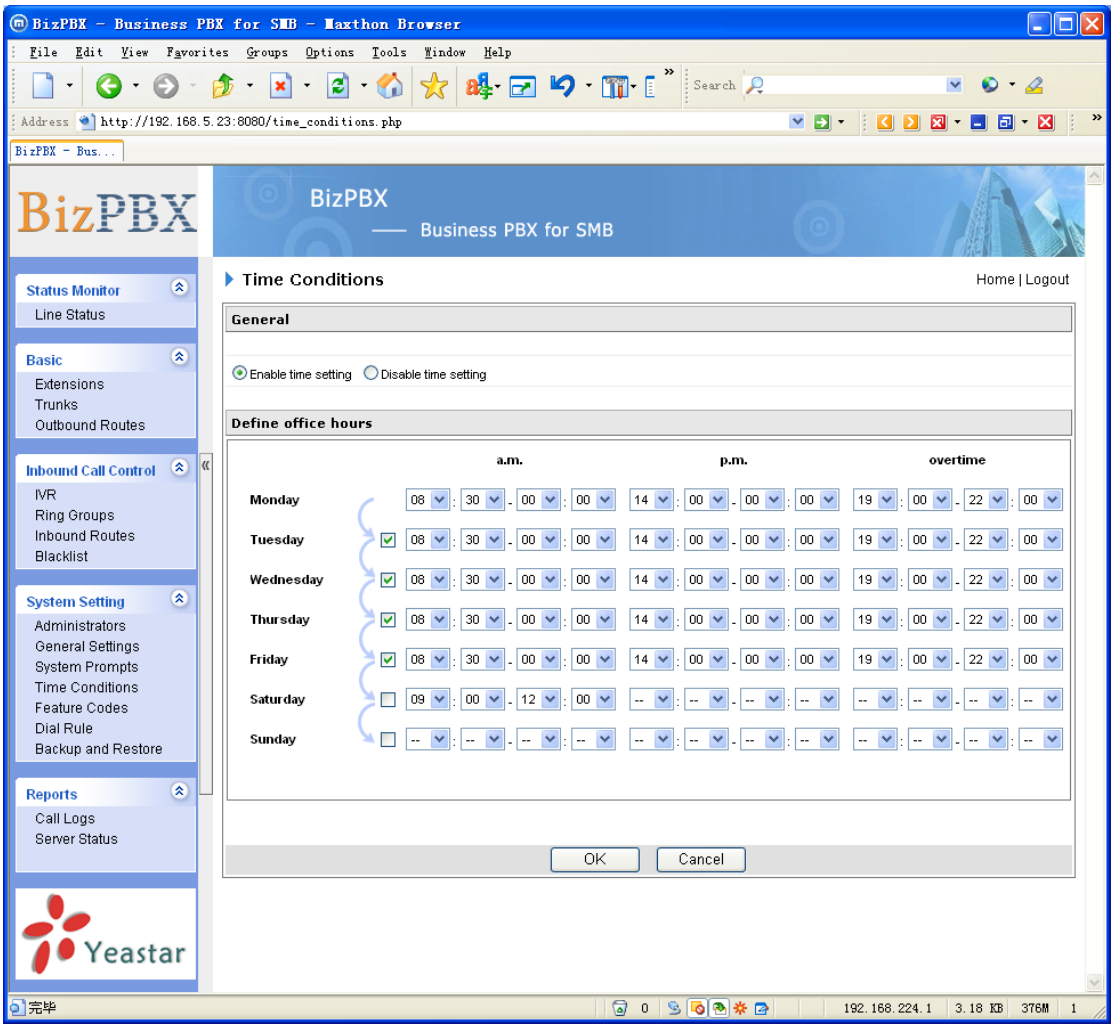


Figure 7-23

7.6.4 Feature Codecs

1）Park current call

Default setting is “**1”

You can press **1 on telephone to hold this call

2）Pick up call from appointed extension

Default setting is “*70”

When Ext.502 is ringing, you can press *70502 on Ext.501 to answer the call.

3）Pick up call from any extension

Default setting is “*71”

When Ext. 501，502，……505 are ringing, you can press *71 to answer

4）Transfer call

Default setting is “**2”

When Ext. 501 and 502 are busy, you can transfer the incoming call to Ext.503 by press **2503

5）Invite to conference

Default setting is “**3”

If Ext.501 wants to invite Ext. 503 into conference, he can press **3503

6）Monitor call from appointed extension

Default setting is “*72”

If Ext.501 wants to monitor Ext. 503, he can press *72503

7）Monitor/Pickup call from appointed extension

Default setting is “*73”

When 502 is in conversation or ringing, Ext.501 can press *73502 to Monitor or Pickup the call.

Note: this function can compatible with Yealink’s T28 IP Phone.

8）Push-To-Talk(PTT)

Default setting is “*74”

When Ext. 501 press *74502, Ext. 502 will answer the call automatically.

Note: this function can compatible with Yealink’s T28 IP Phone.

9）Start Recroding

Default setting is “**4”

When Ext. 501 and 502 are in conversation, he can press **4 to record

10）Stop Recording

Default setting is “**5”

Press **5 to stop the recording

11）Listen to voicemail

Default setting is “*75”

Press *75 to listen to voicemail

12）Pick up current voicemail call

Default setting is “*76”

When someone is leaving a voice mail to Ext.501, Ext.501 can press *76 to answer the call

13）Play previous voicemail

Default setting is “4”

14）Repeat current voicemail

Default setting is “5”

15）Play next voicemail

Default setting is “6”

16）Delete current voicemail

Default setting is “7”

17）Receive FAX

Default setting is “**0”

When someone send FAX to Ext.501, he can press **0 to receive

18）Send FAX to appointed extension

Default setting is “*77”

When someone want to send FAX to Ext.501, he can press *77501.

Figure 7-24

7.6.5 Dial Rule

Making calls through Skype, one have to conform to the Skype dialing scheme as well as calling through PSTN. Maybe you had already got used to PSTN scheme and not accustomed to Skype rules. Therefore, Dial Rule settings will assist you to make Skype calls based on traditional PSTN calling habits. Keep the dialing habit as same as PSTN.

For example, a PSTN user in the mainland of China wants to make an international call IDDD (international direct distance dialing), which format is **00+country code+area code+telephone number**; make a domestic call DDD (domestic distance dialing), which format is **0+area code+telephone number**; While the SkypeOut format, no matter of domestical or international calls, is **00+country code+area code+ telephone number**.

An International call:

1) A Chinese user calls to Canada in traditional PSTN format: the country code is 1, the area code is 416, phone number is “12345678”, then he would dial: **00** +1+ 416 + 12345678;
2) A Chinese user calls to Canada in SkypeOut format: the country code is 1, the area code is 416, phone number is “12345678”, then he would dial: **00** +1+ 416 + 12345678;
In this example, the PSTN and SkypeOur format for an international call is the same.

A Domestic Distance call:
3) A Chinese user in Paking calls to Shenzhen in traditional PSTN format: the area code is “755”, phone number is “12345678”, then he would dial: **0**+755+12345678;
4) A Chinese user in Paking calls to Shenzhen in SkypeOut format: the area code is “755”, phone number is “7571234”, then he would dial: **00+86** (country code)+755+12345678;
In order to not change the PSTN dialing format, this user can take advantage of the Dial Rule to place 0086 with 0, as shown on Figure 49

When the number begins by 00, BizPBX will identify it as international call requirement. When the dialing call begins by 0, BizPBX will identify the 0 as domestic call requirement and transfer it to 0086 automatically to conform SkypeOut format.
Why we substitute 00 for 00? Say it simply, to prevent 00 matches with 0. Because both the international and domestic calls are begin by 0, for fear of replacing 00141612345678 with 008600141612345678, substitution rules priority are listed basipetally. Dialing number will search the suitable rule from the beginning.

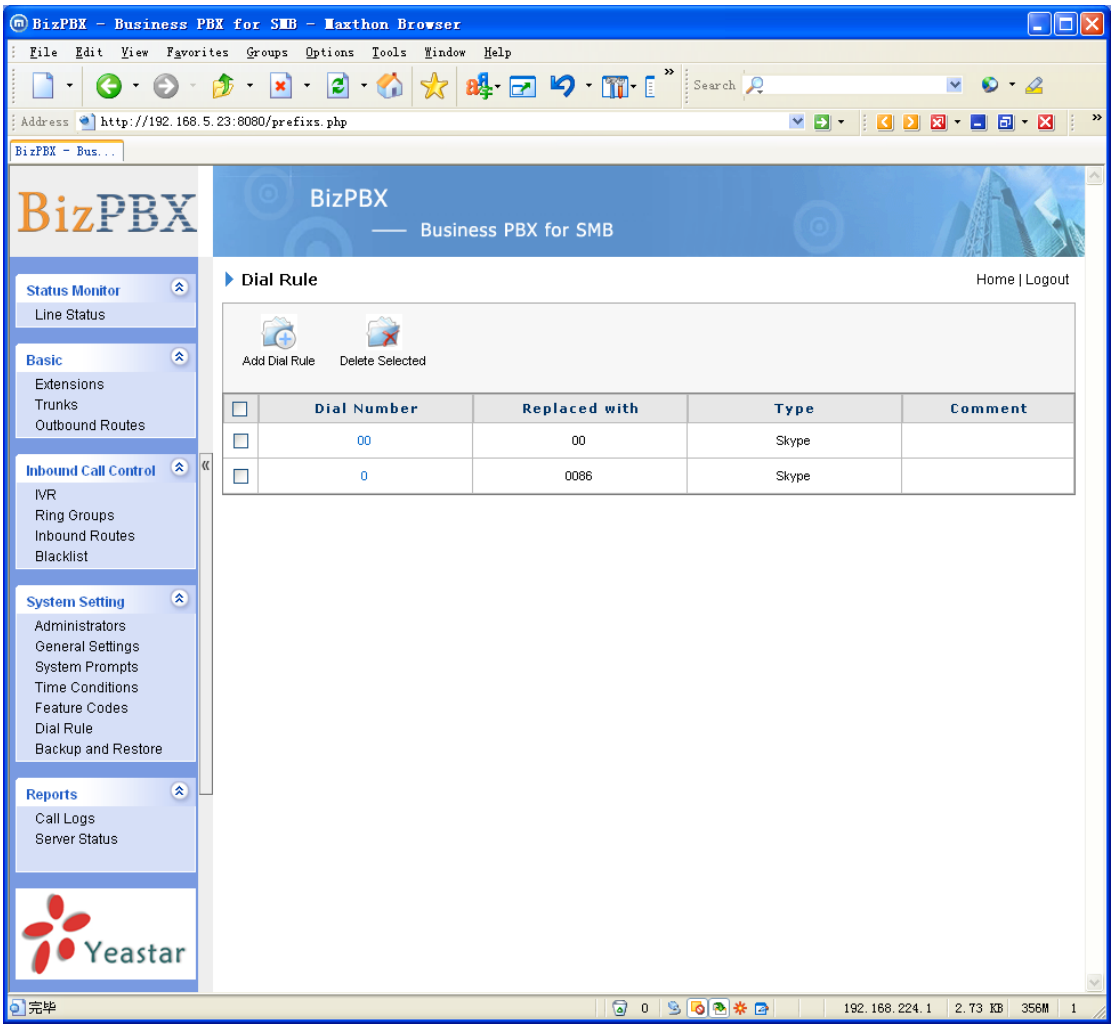


Figure 7-25

Although the Dial Rule Configuration seems a little complicated, you will find its powerful functions if you can well understand its usages.

7.6.6 Public Speed-dial

Admin can maintain the public Speed-dial.

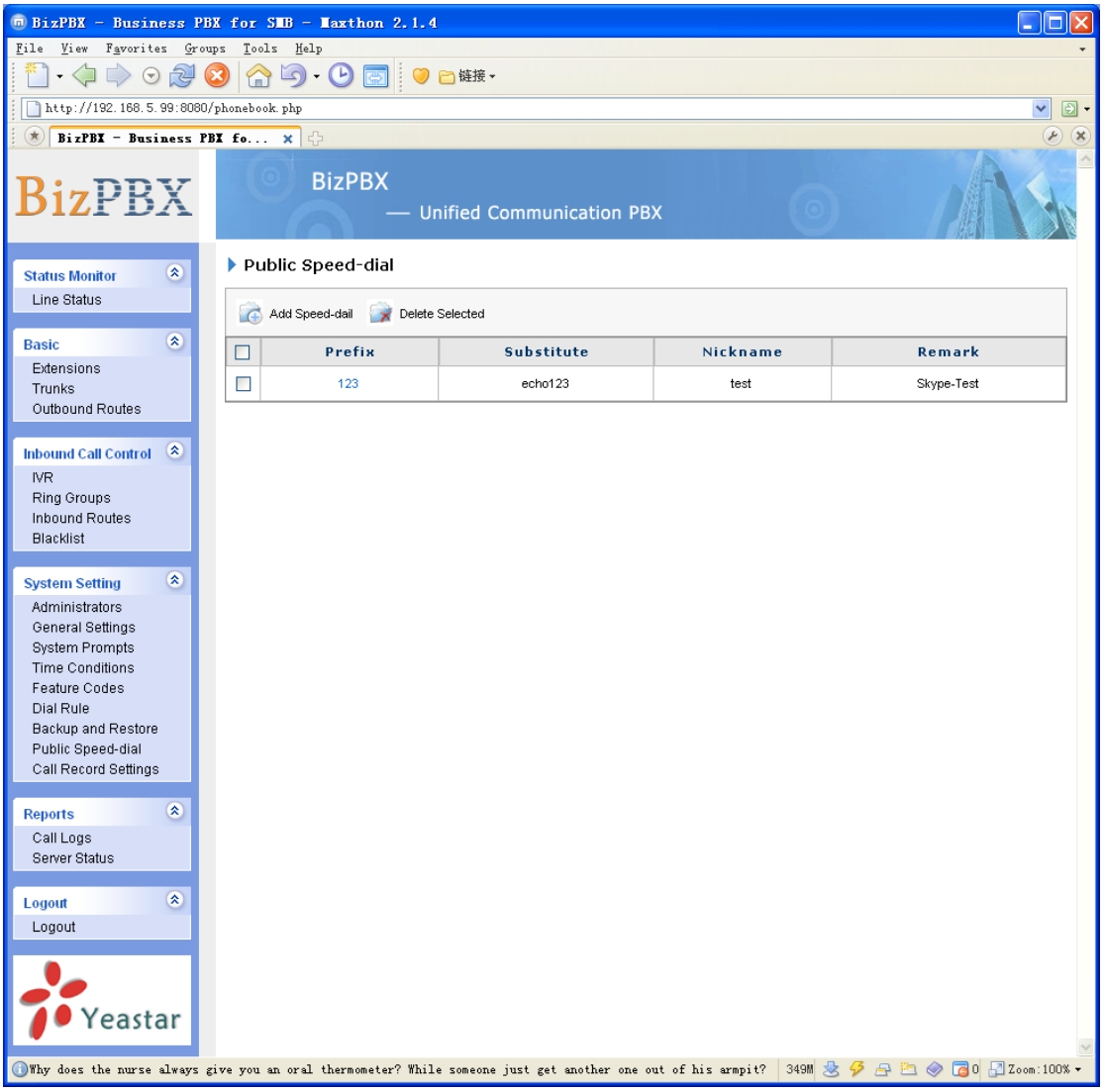


Figure 7-26

7.6.6.1 Add public Speed-dial

1）Click ‘Add Speed-dial’ into manage page.

2）You must enter the ‘ Dial number’ and ‘Skype Account/telephone number’, the rest are options.

Note: If you want to enter a telephone number, please abide to the SkypeOut required format, 00+ country number + city number + telephone number or 00+ coutry number + mobile phone number

7.6.6.2 Delete public Speed-dial

Choose the item and click ‘Delete this item’.

7.6.7 Advanced Module Settings

Administrator can enable the advanced functions of BizPBX that includes automatic recording, desktop monitor and short message service. Just log on BizPBX Client with the authorized extension number, advanced functions are available. Please refer to chapter 7.4.1

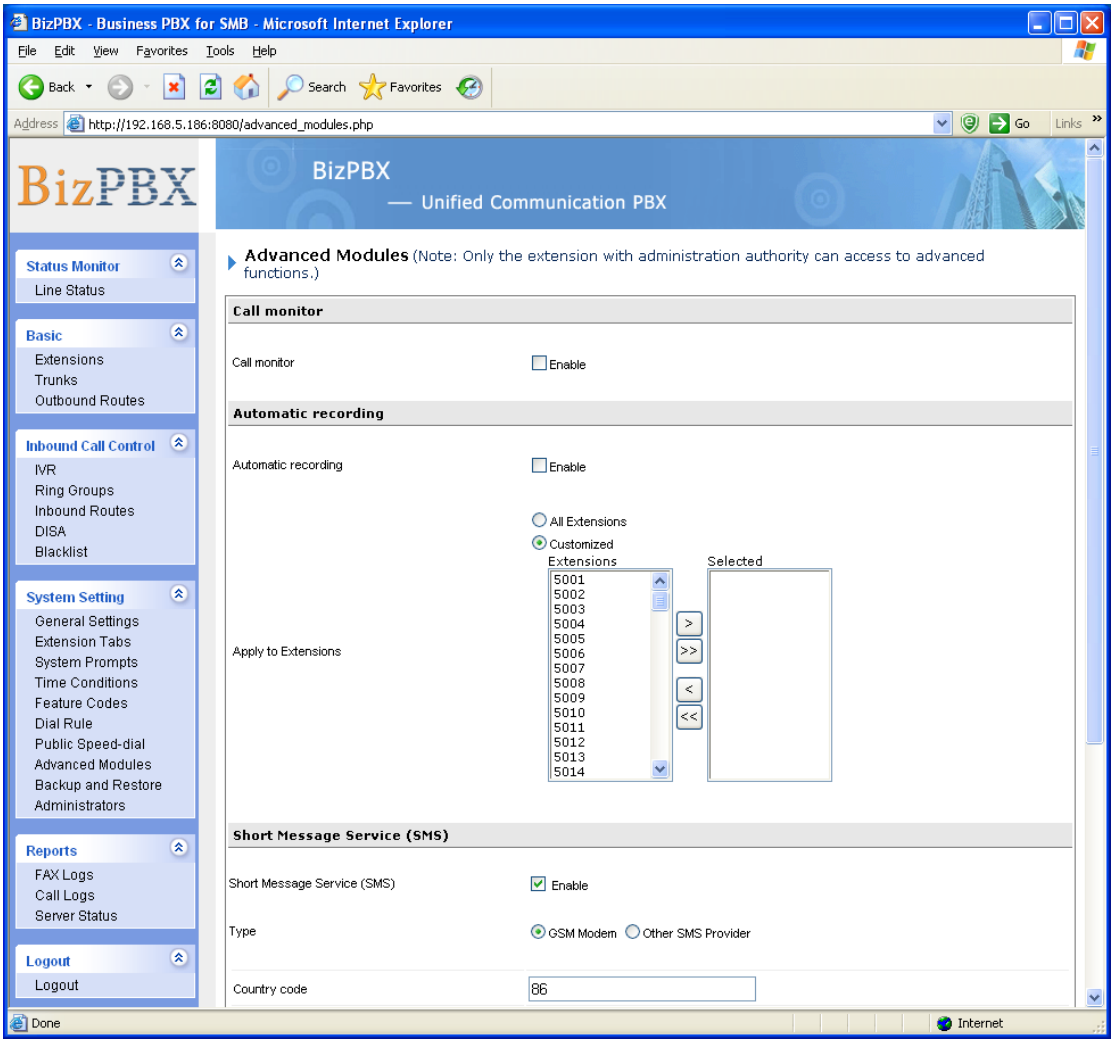


Figure 7-27

7.6.8 Backing up & Restore

As an administrator, you can backup database shown on Figure 46. Click 'Back Up' butoon and choose a destination to save the file.

The **Restore** option will restore your data to the existing database. Select the type of data you want to restore and click **Browse** to choose the location of the backup file to load, and then click **RESTORE**. The data is restored to the database.

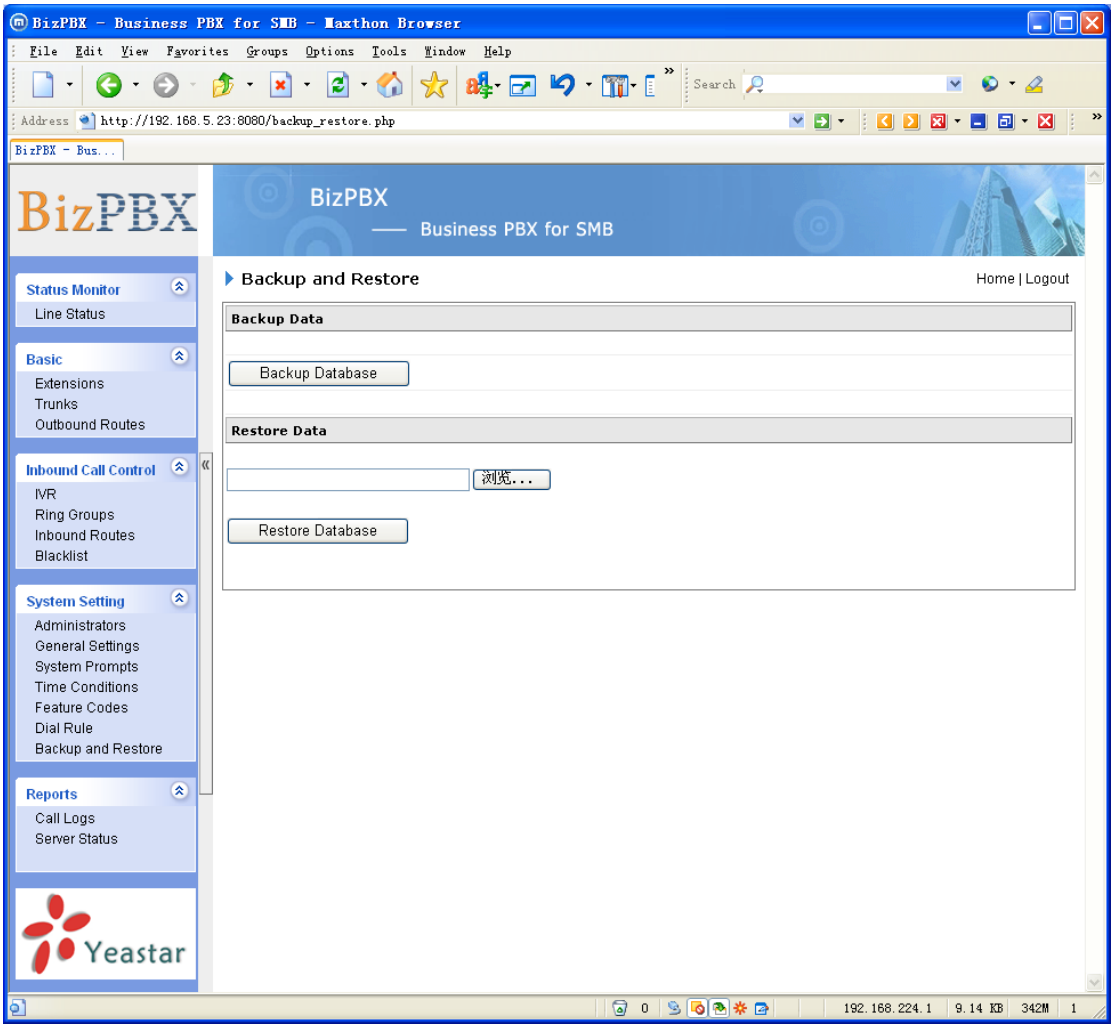


Figure 7-28

7.6.9 Administrator

You can change the password here.

7.7 Reports

7.7.1 FAX Logs

FAX Log captures all call details: Date, Extension, Number, Type, Status and so on. For administrator's convenience, admin can search information by calling time, call type and call duration and export them into CSV file.

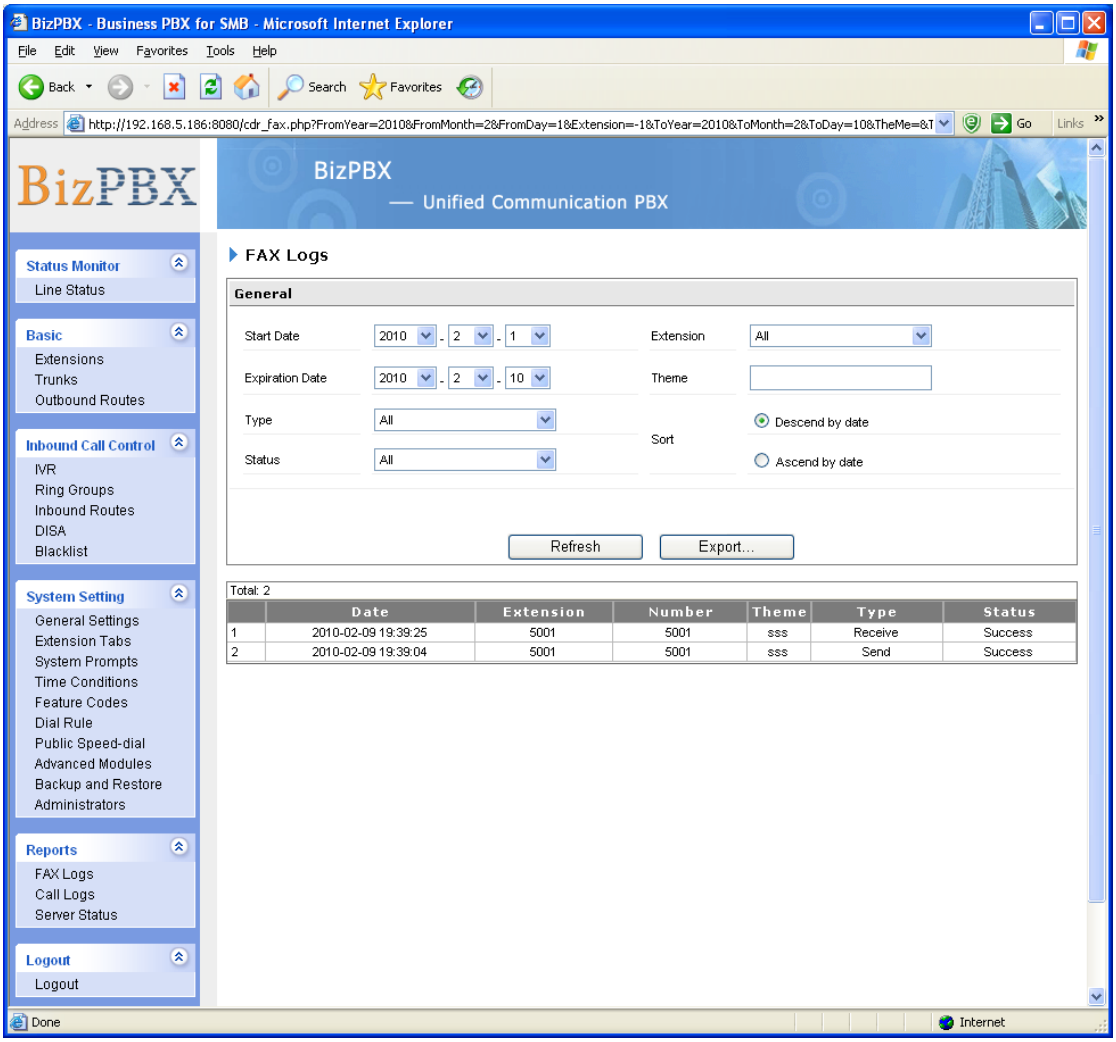


Figure 7-29

7.7.2 Call Logs

Call Log captures all call details: calling time, caller number, callee number, call type, call duration and so on. For administrator's convenience, admin can search information by calling time, call type and call duration and export them into CSV file.

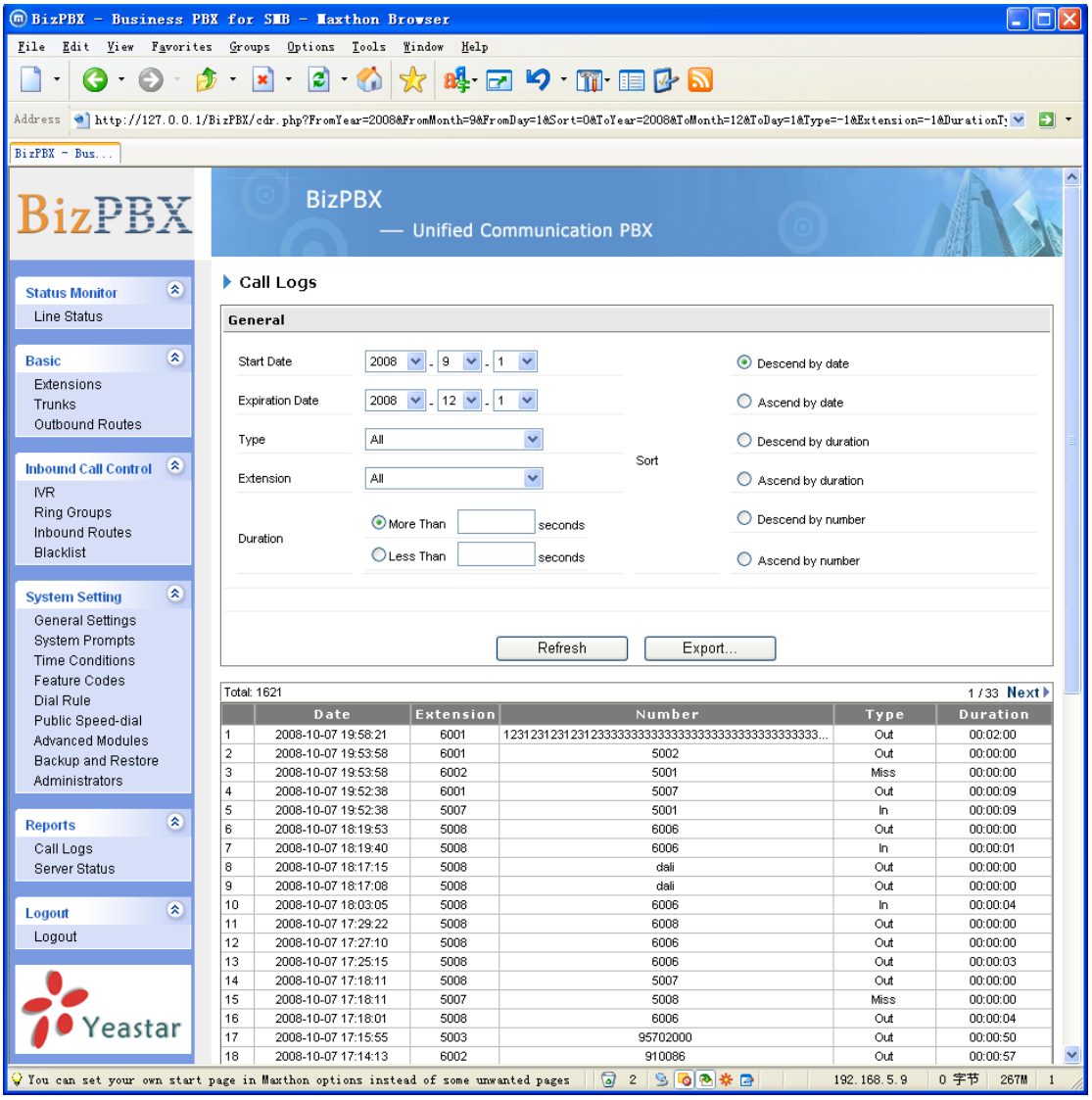


Figure 7-30

7.7.3 Server Status

Administrator can monitor the server performance through staturs.

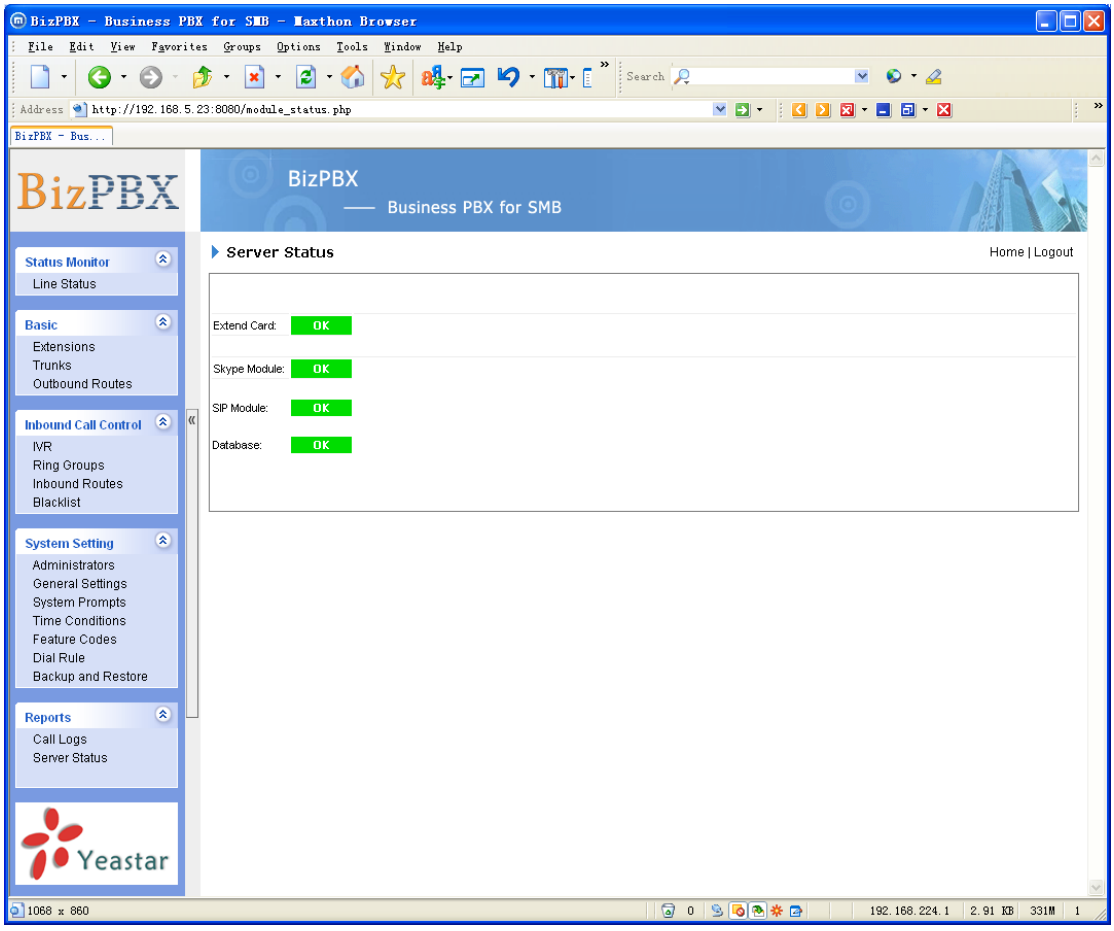


Figure 7-31

8. Using BizPBX

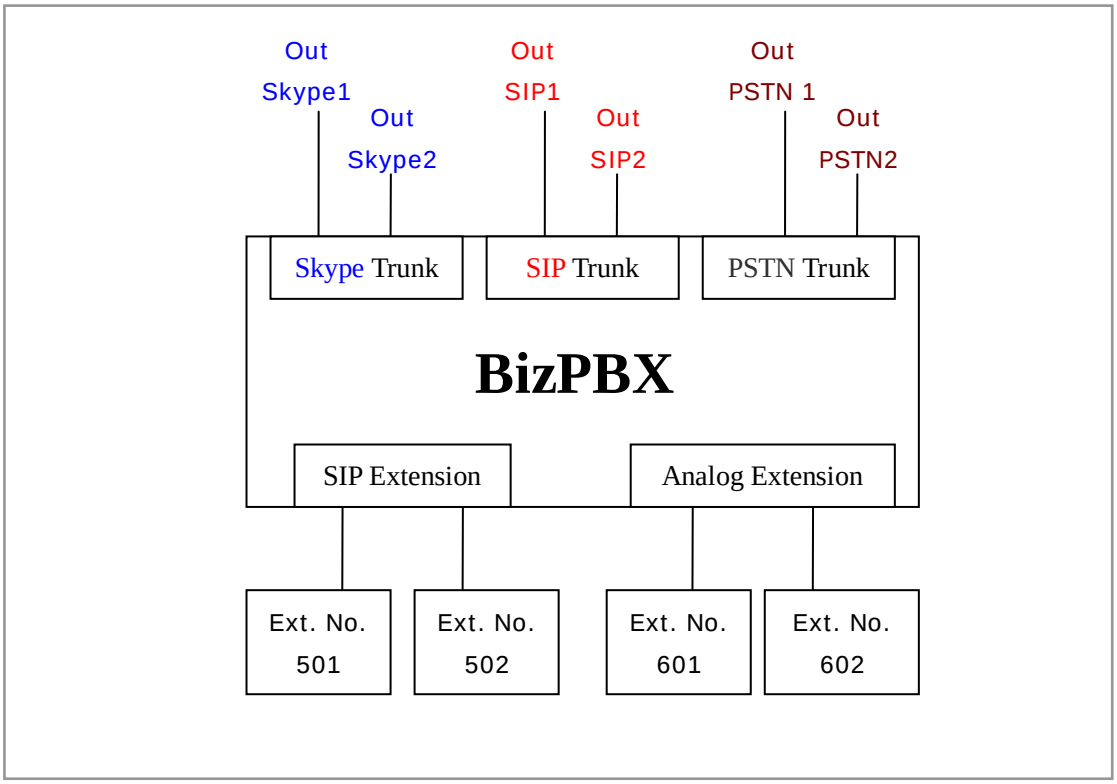
Make three examples for the below typical applications:

- Usage of making outging call
- Usage of answering incoming call
- Website Click-to-Call (SkypeMe) and receving SkypeIn calls

All the telephone numbers are overed by "#" in order to quick up the callout, which is not necessary.

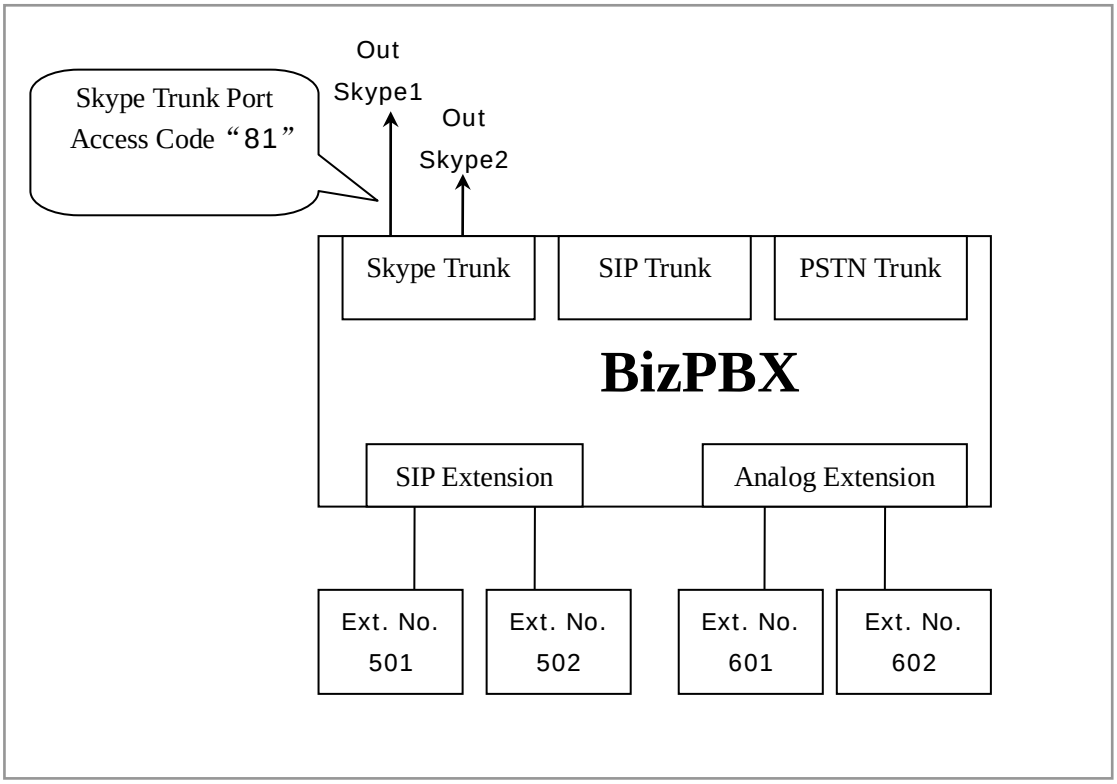
FXO port of TDM800 card should connect with PSTN line, FXS port of TDM800 card should connect with analog regular phone.

For example, BizPBX has SIP Extension 501 and 502, analog Extension 601 and 602, Skype Trunk OutSkype 1 and OutSkype2; SIP Trunk OutSIP1 and OutSIP2; PSTN Trunk OutPSTN1 and OutPSTN2.



8.1 Usage of Making Outging Call

8.1.1 Call Out through Skype Trunk



On the above demonstrated environment:

Suppose the settings of outbound routes are as below:

Route Name：81

Apply this rule to: Calls from extension(s)——All Extensions

Make outbound calls on: Trunk Sequence—— All Skype trunks

Other: Deleted Prefix Count——2

Making calls through SkypeOut:

For example:

1) If a Chinese user (User of Ext. 501) wants to make an international call to Canada through SkypeOut, the number is 001-416-12345678, he can dial: 8100141612345678#;

2) If a Chinese user (User of Ext. 501) wants to make a domestic long distance to Shenzhen through SkypeOut, the number is 755-12345678, suppose he had already setup the Dial Rule as Figure 51 (0 substitiute for 0086): He can dial 81-0-755-12345678#

Here, system will delete the two digits of prefix 81 before sending the signal to Skype.

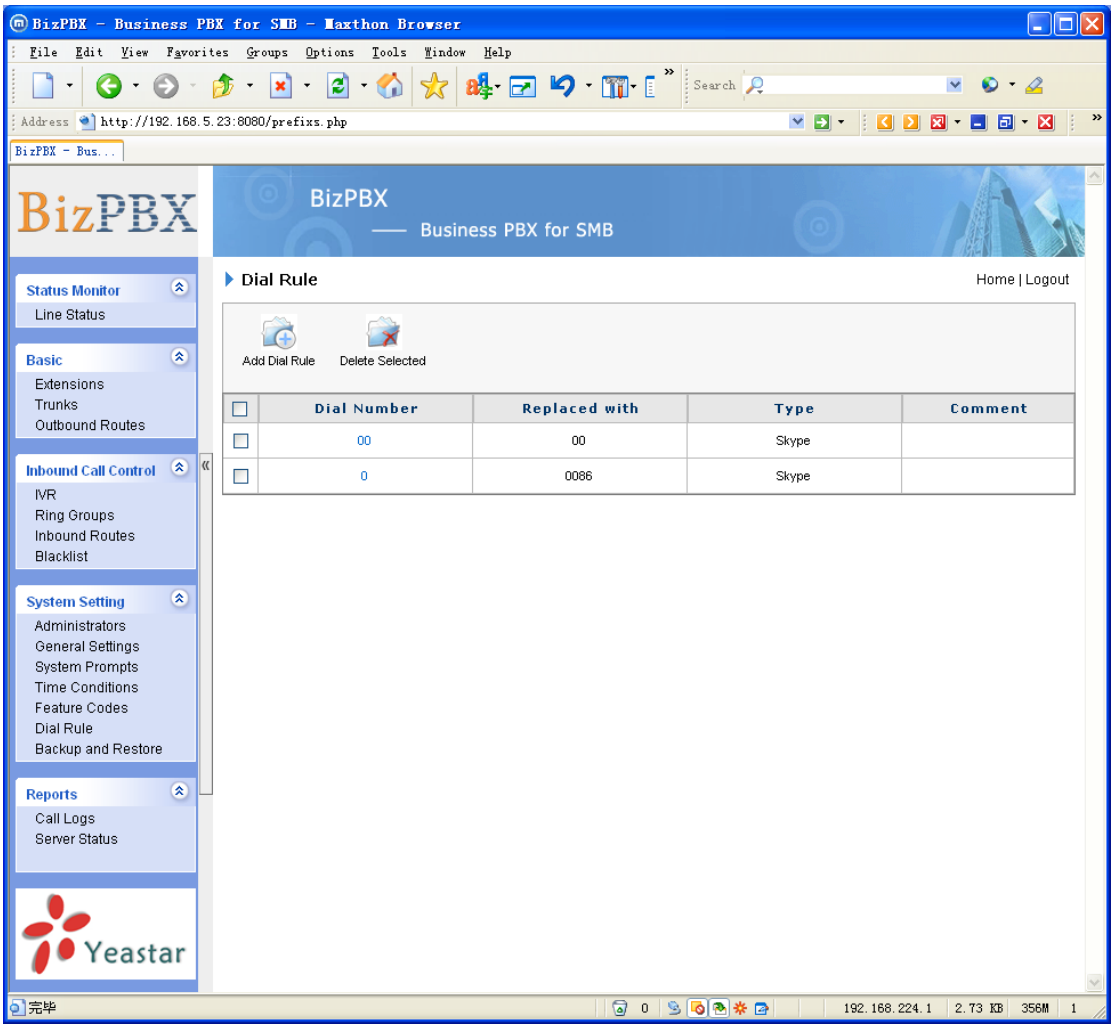
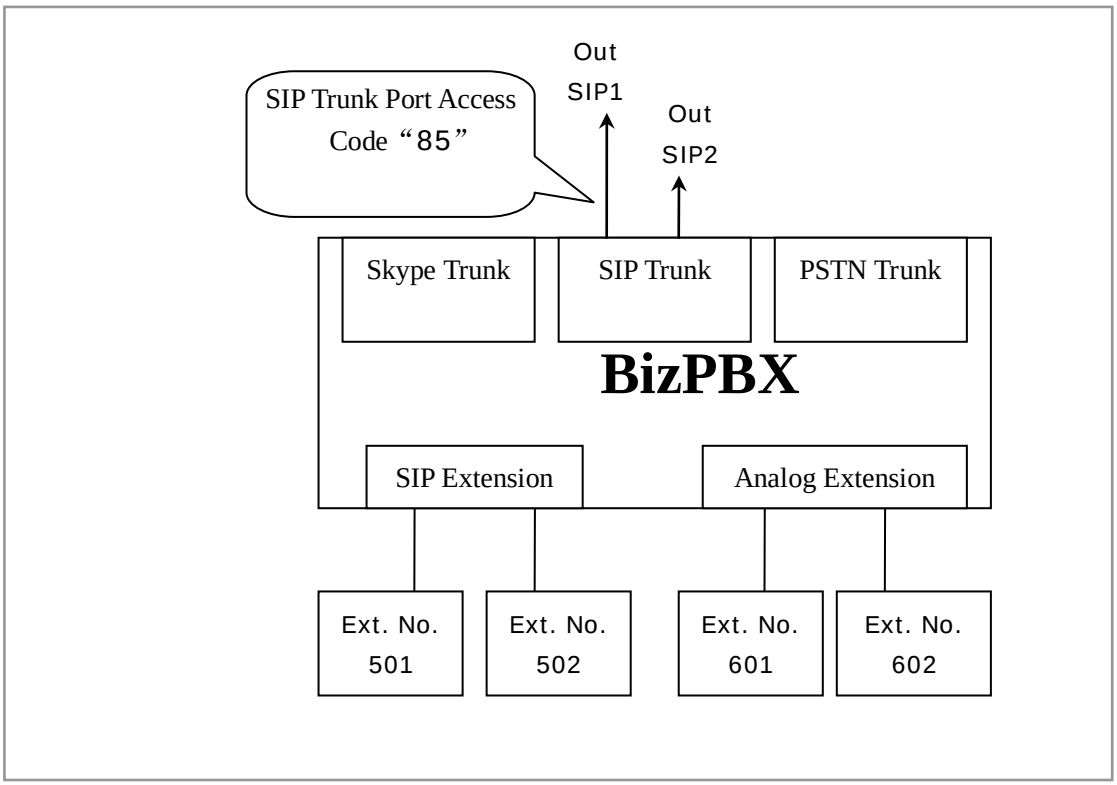

BizPBX - Business PBX for SMB - Maxthon Browser
File Edit View Favorites Groups Options Tools Window Help
Address http://192.168.5.23:8080/prefixs.php
BizPBX - Bus...
BizPBX
Business PBX for SMB
Status Monitor
Line Status
Basic
Extensions
Trunks
Outbound Routes
Inbound Call Control
IVR
Ring Groups
Inbound Routes
Blacklist
System Setting
Administrators
General Settings
System Prompts
Time Conditions
Feature Codes
Dial Rule
Backup and Restore
Reports
Call Logs
Server Status
Yeastar
Dial Rule
Home | Logout
Add Dial Rule
Delete Selected
Dial Number
Replaced with
Type
Comment
00
00
Skype
0
0086
Skype
完毕
192.168.224.1
2.73 KB
358M

Figure 8-1

8.1.2 Call Out through SIP Trunk



On the above demonstrated environment:

Suppose the settings of outbound routes are as below:

Route Name：85

Apply this rule to: Calls from extension(s)——All Extensions

Make outbound calls on: Trunk Sequence—— All SIP trunks

Other: Deleted Prefix Count——2

Making calls through SIP:

For example:

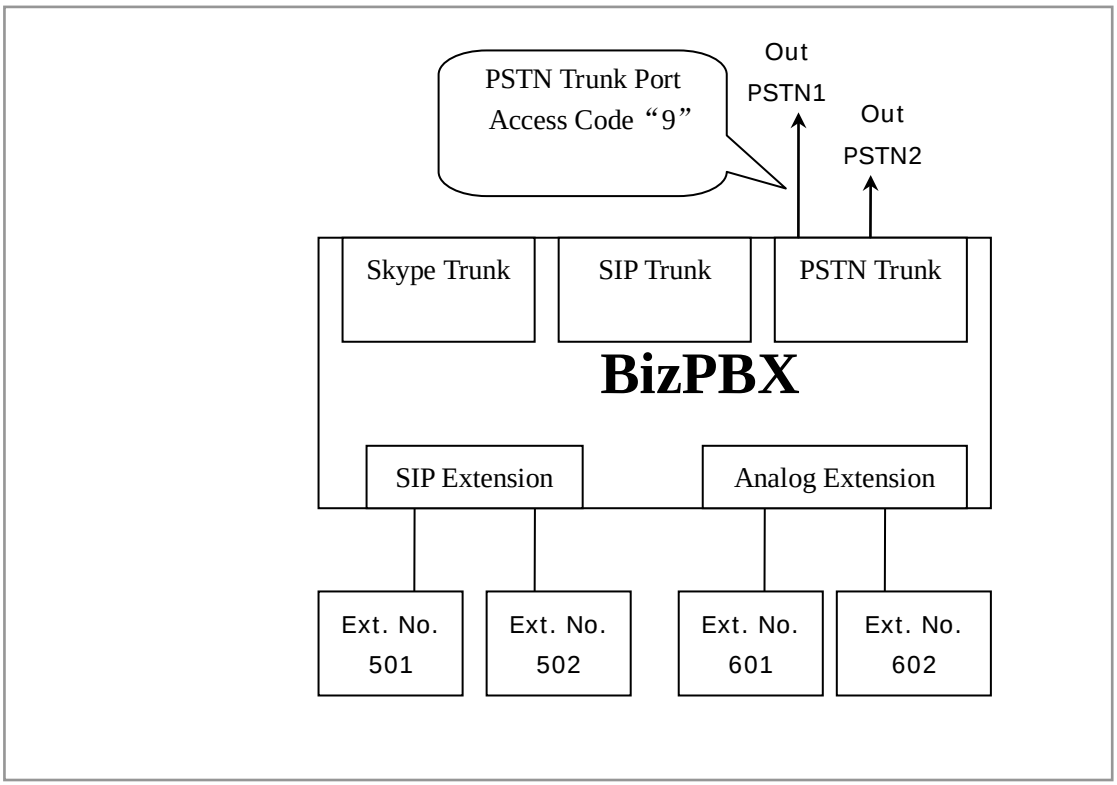
1) If a Chinese user (User of Ext. 501) wants to make an international call through SIP, the number is 001-416-12345678, he can dial: 85001416123456 78#;

2) If a Chinese user (User of Ext. 501) wants to make a domestic long distance call through SIP, the number is 755-12345678, suppose he had already setup the Dial Rule as Figure 51 (0 substitiute for 0086): He can dial 85-0-755-12345678#

Here, system will delete the two digits of prefix 85 before sending the signal to SIP.

Note: The example here, Chinese SIP Provider requires to add country code 0086 before normal numbers. Different SIP providers in different countries have different definitions.

8.1.3 Call Out through PSTN (FXO) Trunk



On the above demonstrated environment:

Suppose the settings of outbound routes are as below:

Route Name：9

Apply this rule to: Calls from extension(s)——All Extensions

Make outbound calls on: Trunk Sequence—— All PSTN trunks

Other: Deleted Prefix Count——1

For example:

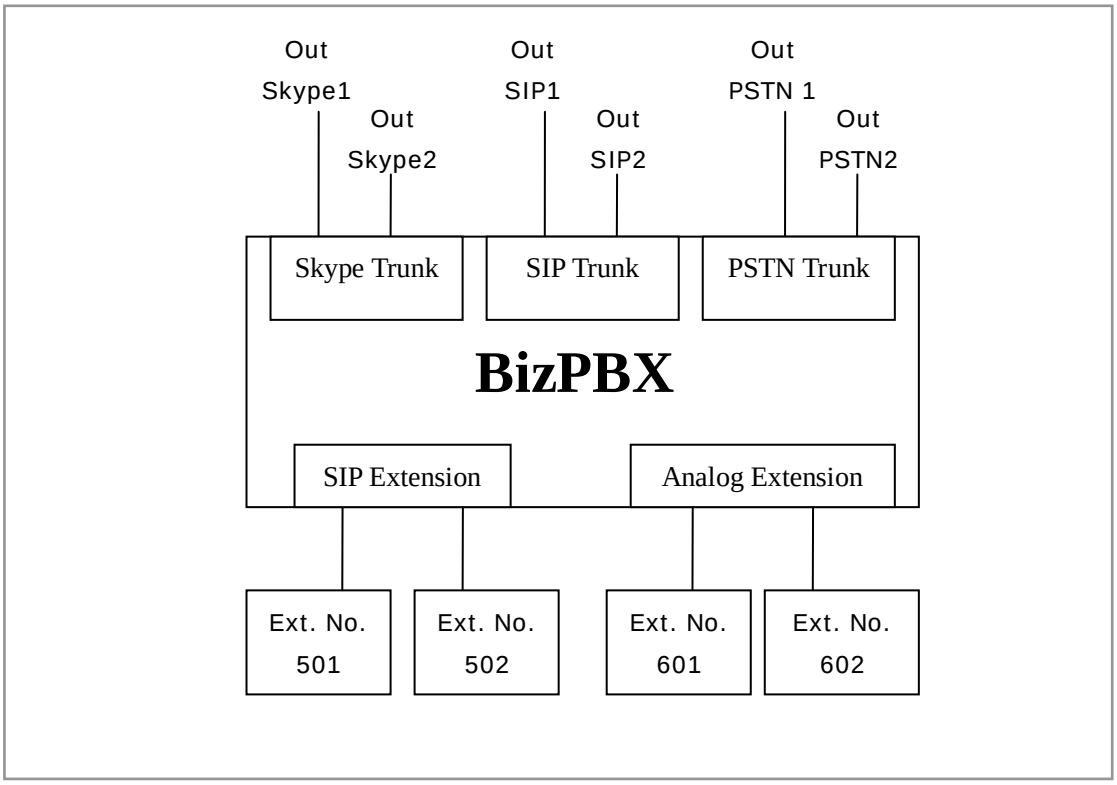
1) If a Chinese user (User of Ext. 501) wants to make an international call through PTSN, the number is 001-416-12345678, he can dial: 900141612345678#;

2) If a Chinese user (User of Ext. 501) wants to make a domestic long distance call through PSTN, the number is 755-12345678; he can dial 9-0-755-12345678#

Here, system will delete the one digit of prefix 9 before sending the signal to PSTN.

8.2 Usage of Answering Incoming Call

8.2.1 Inbound



On the above demonstrated environment:

Suppose the settings of inbound routes are as below:

Route Name: All

Source:Trunk Sequence——All Trunks

Route calls during office hours to: Destination——IVR 7777

Route calls outside office hours to: Destination——End Call

For example: All the incoming calls during the office hours will be answered by IVR 7777 and then forwarded to corresponding extension according to the transferring list. And system will end the calls directly if outside of working hours.

8.3 Usages of Website Click-to Call (SkypeMe) & SkypeIn

Using BizPBX, you can make a Company Skype ID through two ways: apply an ID and public it on website directly as shown on figure 52, or apply an SkypeIn number as Company Skype ID, and then clients can make call to company through web directly and multiple concurrent calls are acceptable.

8.3.1 Showing Company Skype ID as SkypeMe on Website

It allows visitors to call the company directly through SkypeMe button.

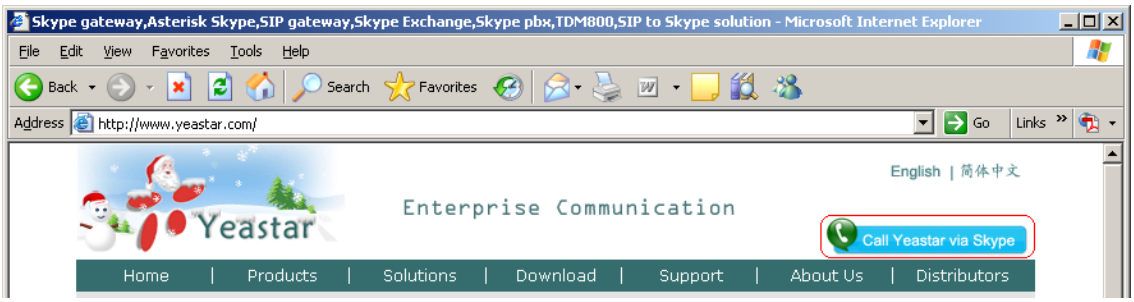


Figure 8-2

Step 1: Allow online status to be shown on the website

- Log in Skype on Company Skype ID (Skype1)
- Click 'Tools' → 'Options' on menu, as Figure 53
- Click on checkbox 'Allow my status to be shown on web' as Figure 54
- Save settings

Step 2: Public SkypeMe button on the website
Please add the following html code on the relevant position of the web:
**

Here *Skype1* stands for your Company Skype ID.

Step 3: Company Skype ID is complete. You are now ready to receive calls from website.

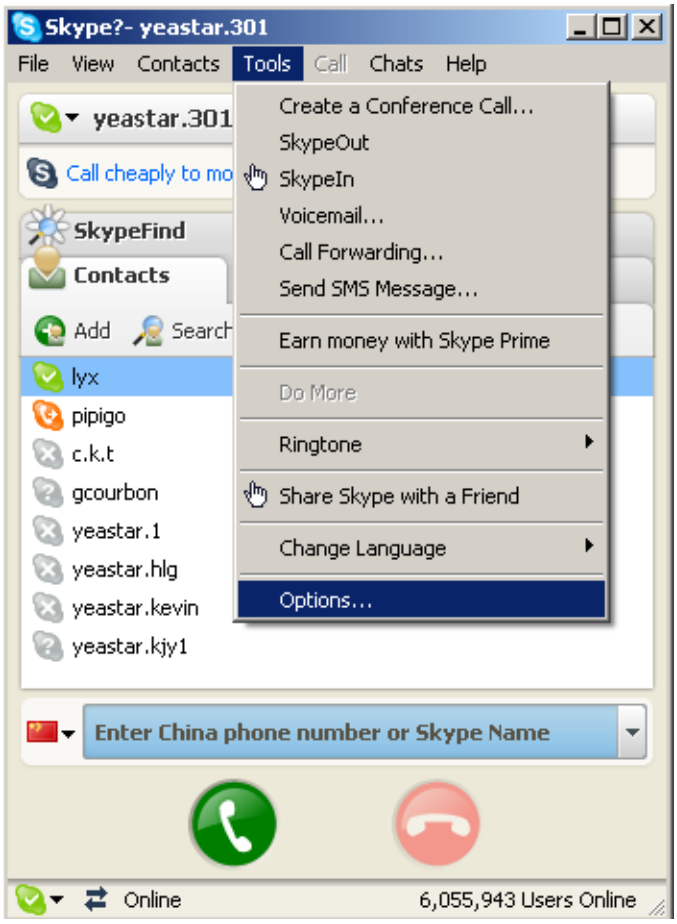


Figure 8-3

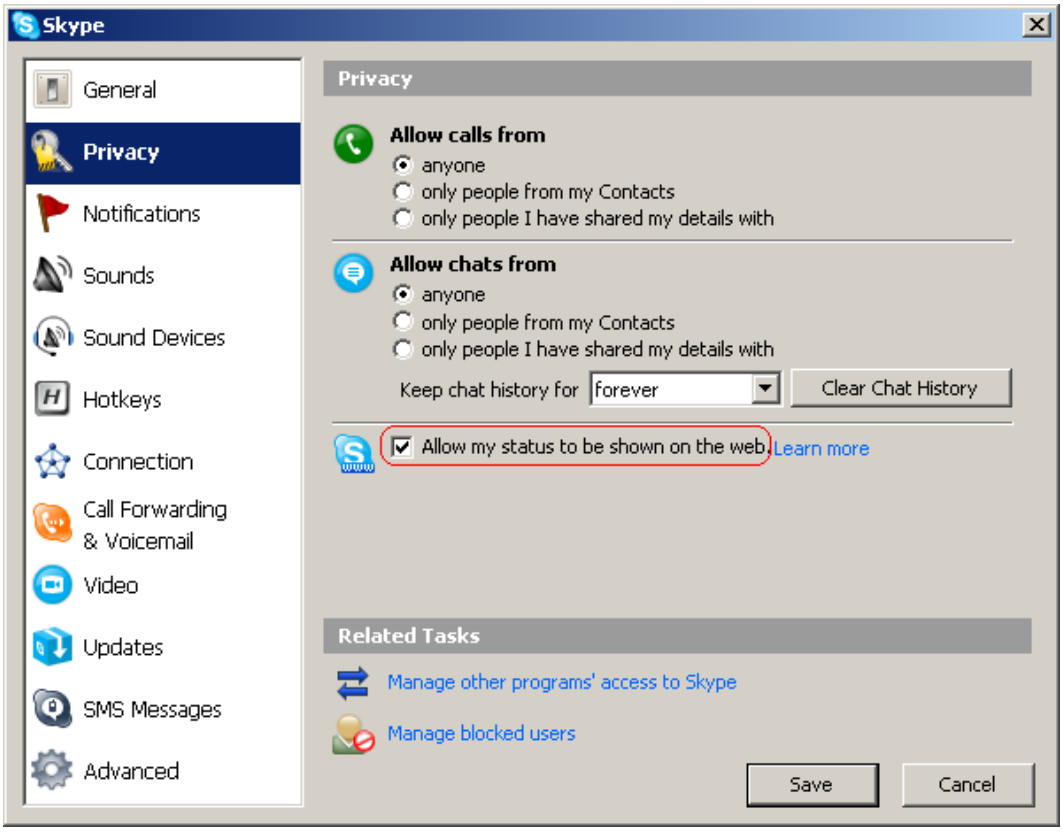


Figure 8-4

9. Uninstalling BizPBX Software

Uninstalling BizPBX Software will delete all your data, including phonebook, call log and recorded voice prompt recordings on this PC. Before you uninstall BizPBX, make sure you have backed up the necessary data.

Method 1：

1. Go to the Window's Start menu and open BizPBX Uninstall, click Yes on the pop-up confirmation window 'Do you want to uninstall BizPBX'.

2. The next prompt window will guide you to Close the current running BizPBX program first. Choose 'Stop' to abandon uninstallation; 'Cancel' to continue the uninstallation program.

3. The Uninstallation process will delete the BizPBX files from your computer. When the uninstallation process is complete, the Uninstall Finished screen will pop up. Click Finish button.

Method 2：

1. Go to the Window's Start menu and open the Control Panel.

2. Go to the Add or Remove Programs, which will open a new window with a list of software programs installed on the computer. Scroll down to BizPBX and click Change/Remove button.

10. Installing TDM800 Card

In order to use traditional telecom PSTN line and connect with analog phone, users need to install **Yeastar's** TDM800 PCI card into BizPBX server computer.

Steps:

1. Power down the PC
2. Unplug the PC from its main AC power supply
3. Plug the hard disk power supply cable to power jack of TDM800 card for FXS port power supply.
4. Fix the TDM800 card and make sure it stays in good contact with PCI slot.
5. Connection: FXS port connects to regular phone; FXO port connects to PSTN line.

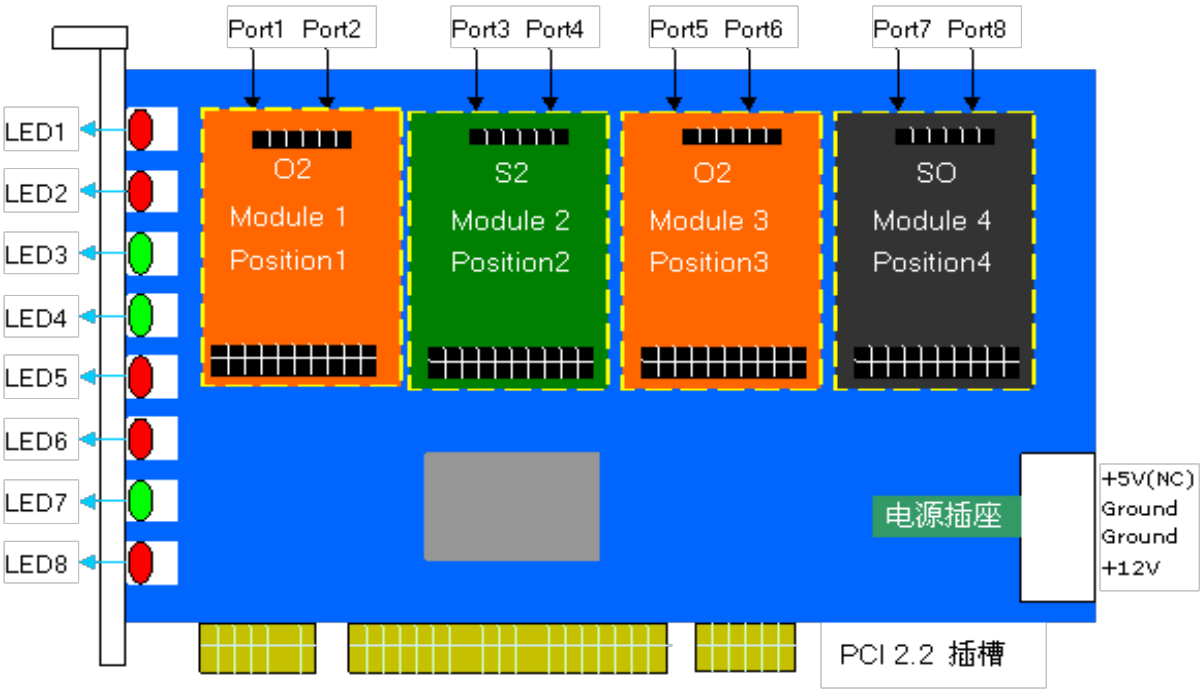


Figure 10-1

TDM800 can install 4 modules of S2 (FXS) and/or O2 (FXO) , each one with 2 ports.

Red light: it stands for the FXO port.

Green light: it stands for the FXS port.

6. Power on the PC

That's all for installation hardware card TDM800.

How to confirm the card is installed successfully?

When continue to section 6 'Run BizPBX Config. Wizard', you will see the lighted LED on TDM800 card. If the Wizard reads all the configurations of card correctly, it indicates the success of installation. Or else, please check if the PCI slot is in good contact. If you are unable to clear of trouble, contact your reseller.

Appendix A Customizing WAV Format （Auto Attendant）

Here, users can customize a WAV file through Windows accessories recorder, or through other software to finish the format, like CoolEdit.

(WAV Format： 8,000 kHz,4 Bit, Mono 3KB/sec)

Step:

Start menu→Program→Attachment→ Entertainment→ Recorder

1. Record the prompt message file

Figure A-1

2. Save it As

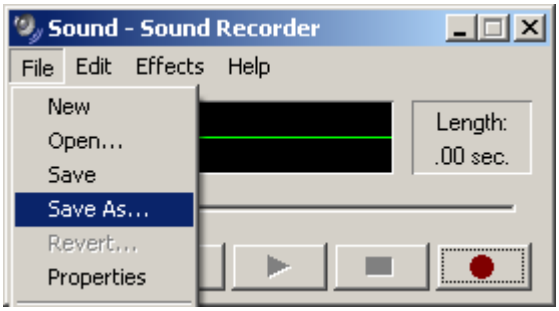


Figure A-2

3. Change the format; click the button as Figure 56

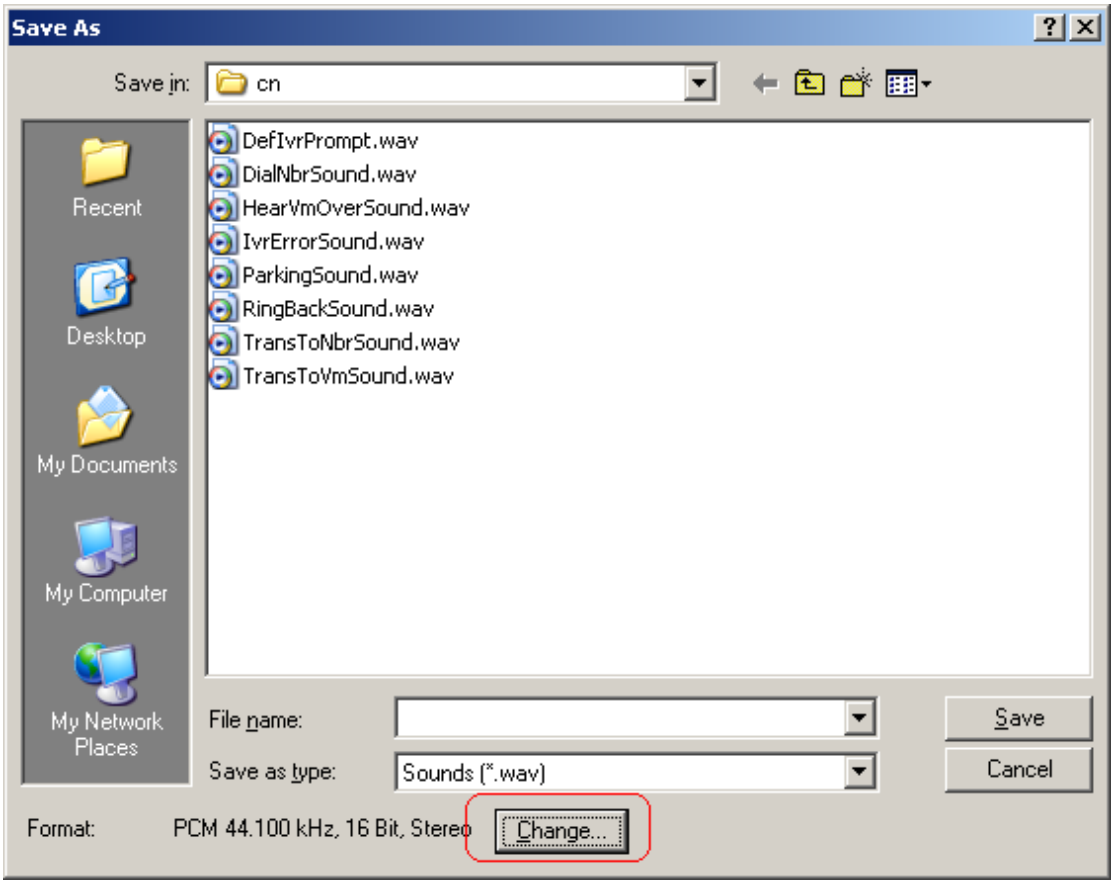


Figure A-3

4. Choose an entry on 'Sound Selection'

Format: IMA ADPCM

Attributes: 8,000 kHz,4 Bit, Mono 3KB/sec

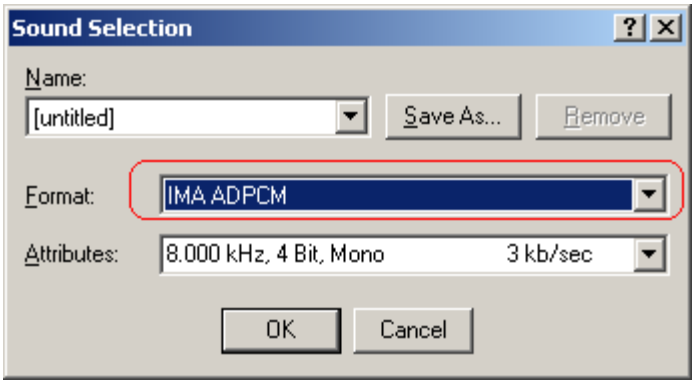


Figure A-4

5. Click 'OK' to save the settings and voice files.

Appendix B Glossary

Busy tone: Low-pitched tone interrupted sixty times per minute. This tone indicates that the called station is off-hook or busy.

Dial tone: A constant high-pitched tone indicating that the system is ready to process your call. It's essential that users listen to this tone before dialing phone numbers.

DTMF（Dual Tone Multi-Frequency）: DTMF is used for most call set ups between the telephone set and telephone switch.

FXO（Foreigh Exchange Office）: FXO interface receives POTS Service, typically from a Central Office of the Public Switched Telephone Network

FXS（Foreigh Exchange Station）: FXS is the interface on a VoIP device for connecting directly to phones, faxes, and CO ports on a PBX or key telephone system.

Off-hook: The handset is removed from the cradle or you are in conversation.

On-hook: The handset is in the cradle.

PBX（Private Branch Exchange）: It is a private telephone network used within an enterprise. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.

PSTN（Public Switched Telephone Network）: It is the world's collection of interconnected Voice-oriented public telephone networks.

VoIP (Voice over Internet Protocol): A category of hardware and software that enables people to use the Internet as the transmission medium for telephone calls by sending voice data in packets using IP rather than by traditional circuit transmission of the PSTN.

Appendix C Country/Region List

BizPBX supports the following countries' and regions' telecommunication parameters:

ARGENTINA
AUSTRALIA
AUSTRIA
BAHRAIN
BELGIUM
BRAZIL
BULGARIA
CANADA
CHILE
CHINA
COLUMBIA
CROATIA
CYPRUS
CZECH
DENMARK
ECUADOR
EGYPT
ELSALVADOR
FINLAND
FRANCE
GERMANY
GREECE
GUAM
HONGKONG
HUNGARY
ICELAND
INDIA
INDONESIA
IRELAND
ISRAEL
ITALY
JAPAN
JORDAN
KAZAKHSTAN
KUWAIT
LATVIA
LEBANON
LUXEMBOURG
MACAO
MALAYSIA
MALTA
MEXICO
MOROCCO
NETHERLANDS
NEWZEALAND
NIGERIA
NORWAY
OMAN
PAKISTAN
PERU
PHILIPPINES
POLAND
PORTUGAL
ROMANIA
RUSSIA
SAUDIARABIA
SINGAPORE
SLOVAKIA
SLOVENIA
SOUTHAFRICA
SOUTHKOREA
SPAIN
SWEDEN
SWITZERLAND
SYRIA
TAIWAN
THAILAND
UAE
UK
USA
YEMEN

Appendix D FAQ

A. Skype Issues

1. **Q: What is the bandwidth for each Skype Call?**
   A: The bandwidth of one Skype Call depends on Internet access bandwidth and PC's CPU performances of the both sides of the conversation. According to statistics provided by the official website of Skype, each Skype Call requires an average bandwidth of 3Kb/s-16Kb/s, please refer to http://www.skype.com/help/faq/technical.html

2. **Q: What is SkypeOut?**
   A: SkypeOut allows you to place calls to regular telephones (landlines or mobiles) all over the world for a low fee.

3. **Q: What is SkypeIn?**
   A: SkypeIn allows you to own a regular phone number. Thus your contacts can call your Skype account on a regular phone.

4. **Q: Can BizPBX work with SkypeIn?**
   A: BizPBX can work with SkypeIn, especially SkypeIn in some areas can support concurrent multiple incoming calls.

5. **Q: What is SkypeMe?**
   A: Please refer to Figure 69. You can make a SkypeMe button on the web or email and allow visitors to view the online status so as to click the button to call the relevant Skype account.

Figure D-1

6. **Q: How can I set my Skype status always 'Online'?**
   A: You need to do nothing for it. BizPBX will set it for you automatically and Skype of all trunks will be kept 'Online' status always.

B. BizPBX Issues

1. **Q: Why the LEDs on card are still not lighted yet after I installed TDM800 driver through running BizPBX Installation Wizard?**

   A: Firstly please make sure whether you cloned the operating system, and then check if hardware TDM800 card stays in good contact with PCI slot.

2. **Q: Why the Skype can't be started automatically when I running BizPBX Installation Wizard?**

   A: Main reasons maybe are:

   1) This PC is Domain Controller.

   2) This PC has minimum password length requirement (no zero).

   3) Services in the **Manage and Applications**, the **Secondary Logon** service haven't been started.

3. **Q: What is the function of the one DOS program that launches after completing the installation of BizPBX software?**

   A: The DOS program is used for Web/SQL services. Please do not close it.

4. **Q: Why can't I make SkypeOut calls?**

   A: Main reasons are: 1) have not purchased SkypeOut credit for the relevant trunk's Skype ID; 2) dialed telephone numbers in error SkypeOut format, the right format is: 00+country code+ area code+ telephone number or mobile phone number

   If you encounter any other problems about our products, please contact us or view in the following way: http://www.yeastar.com/faq.asp

Appendix E Manage E1/T1 DID

After configured E1/T1 hardware card in system, there will be an interface as following:

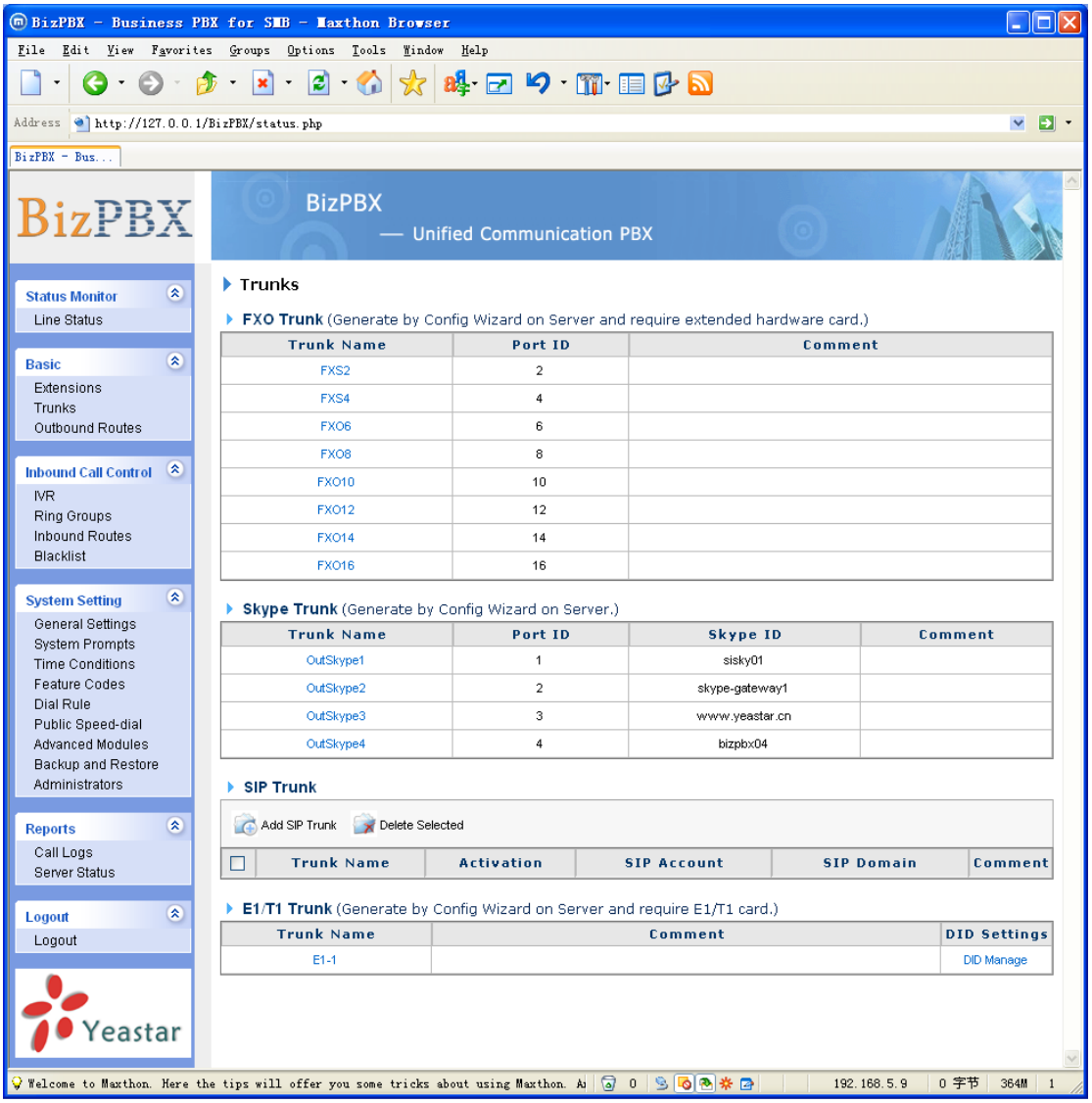


Figure E-1

As on Figure E-1, click 'DID Manage' to access into the next DID manage interface as Figure E-2.

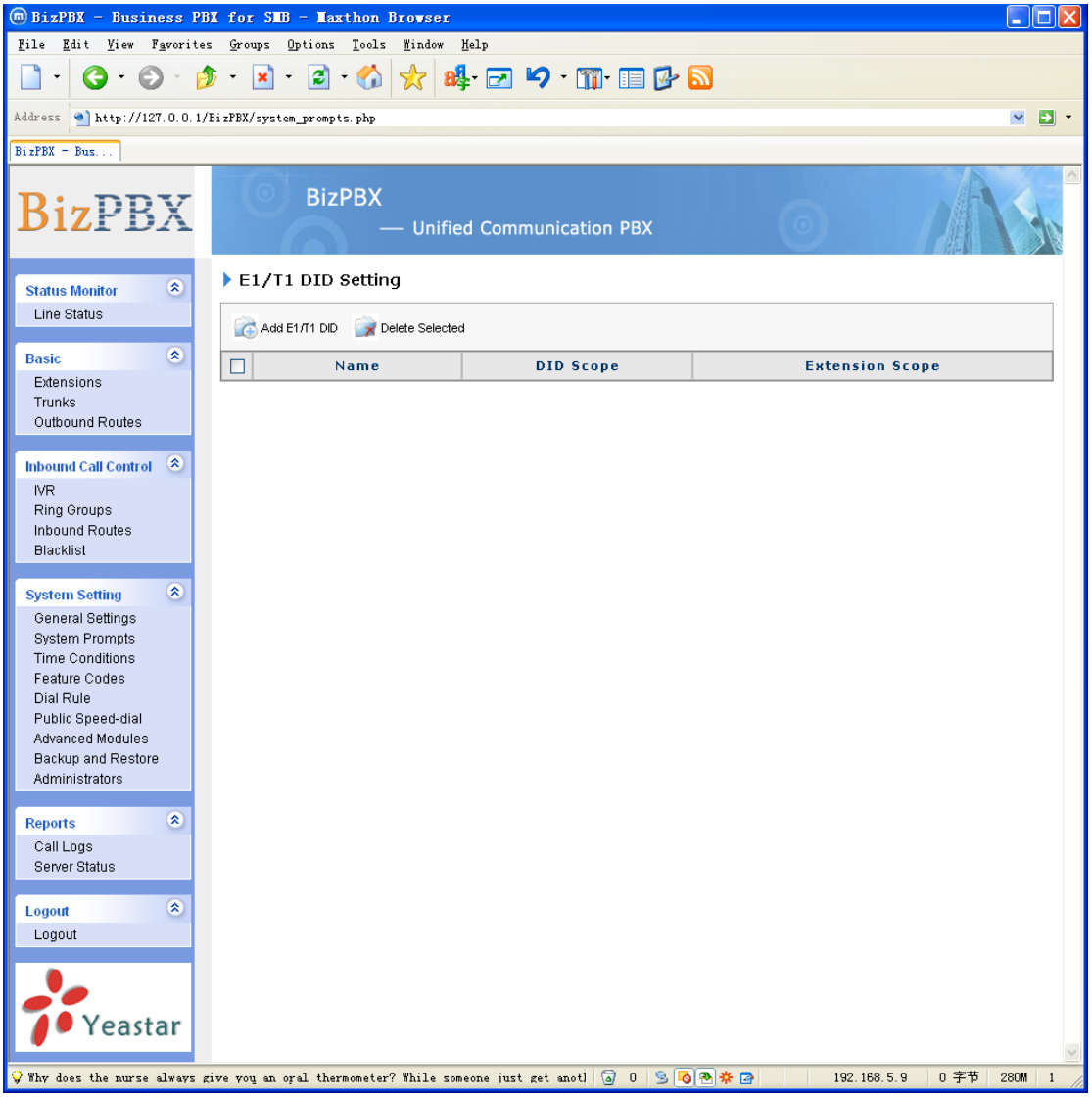


Figure E-2

1. Add E1/T1 DID

You're required to input following information:

·Number – Number of DID

·Begin from DID –Means the initial DID number

·Begin form Extension – Extension number for corresponding DID

For Example, if users filled in following information:

Number: 2

Begin from DID: 5503301

Begin from Extension: 301

Now there are two DID numbers with two extensions:

5503301 corresponds to 301

5503302 corresponds to 302

When calling 5503301, system will auto call 301 extension; it's the same

when calling 5503302, system will auto call 302 extension

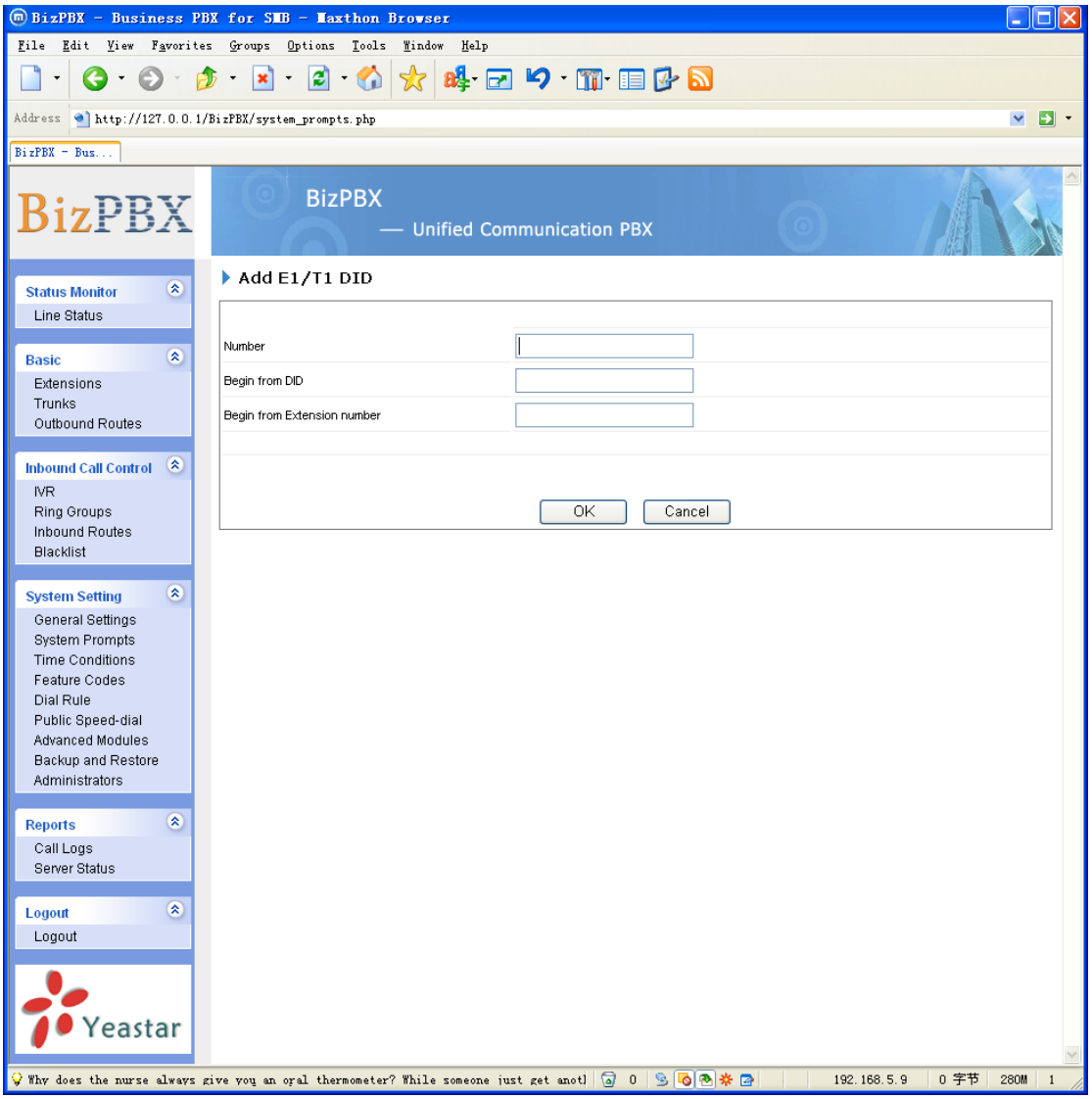


Figure E-3

2. Delete DID

Choose the DID item and click ‘Delete this item’.

Thank you for using BizPBX. We value your comments and concerns.

<Finished>